EXPERIMENT NO-1

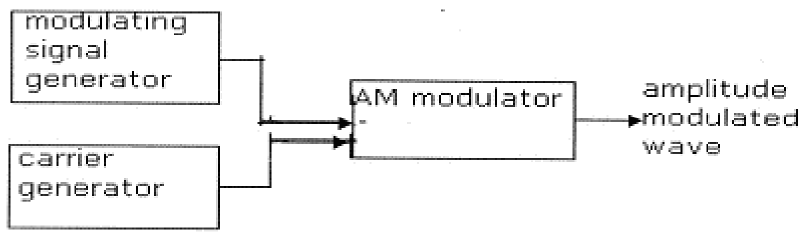
AMPLITUDE MODULATION &DEMODULATION

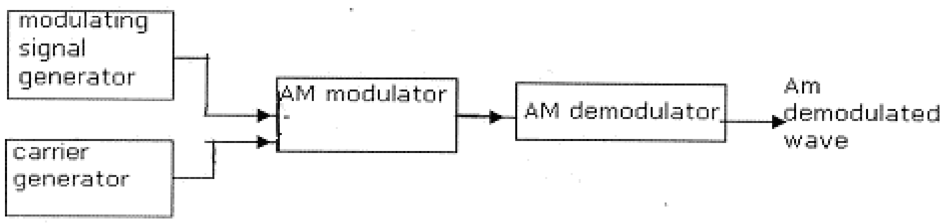
AIM: To study the function of Amplitude Modulation & Demodulation also to calculated the modulation index.

APPARATUS :

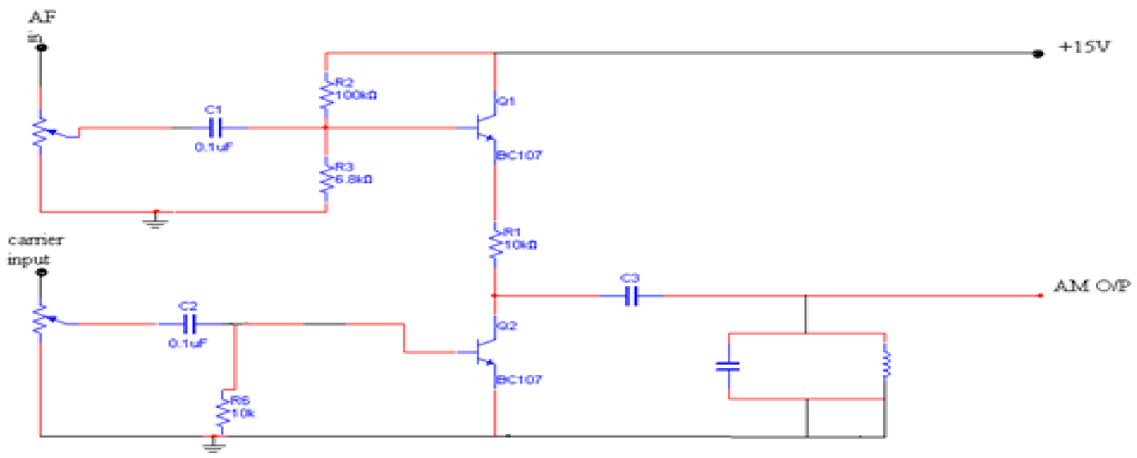
1. Amplitude Modulation & Demodulation trainer kit.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Connecting cords & probes.

BLOCK DIAGRAM:

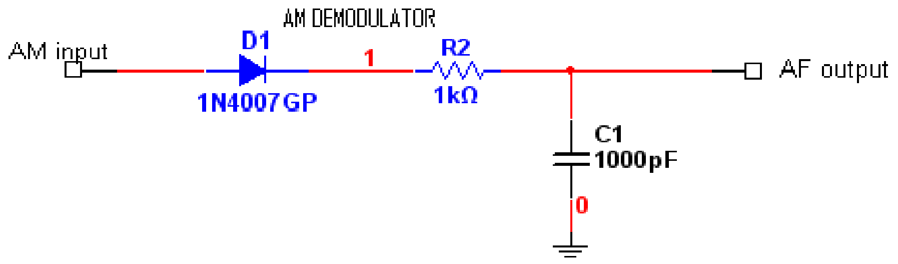


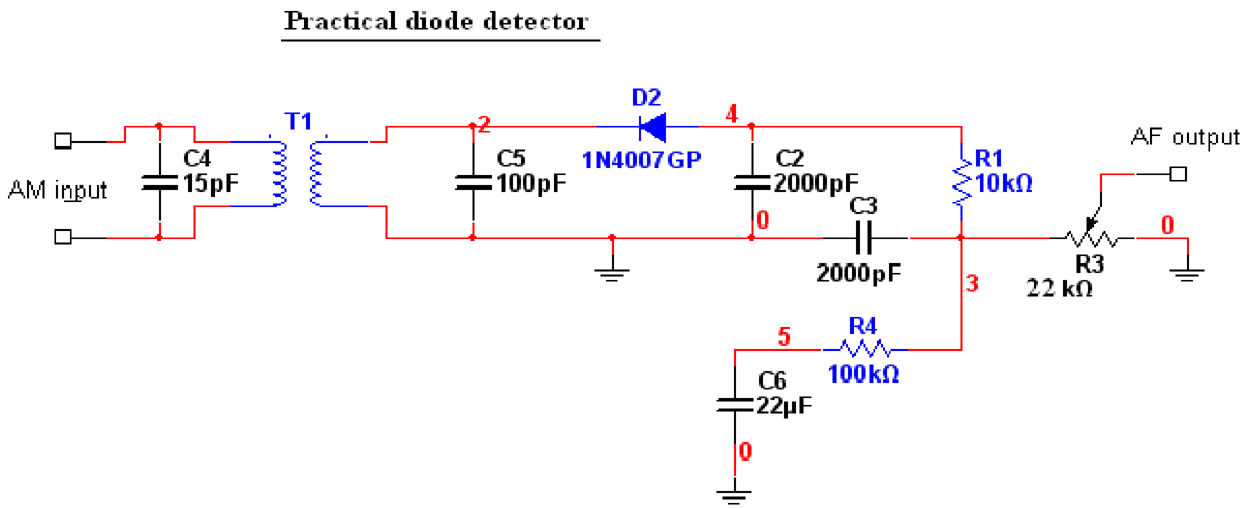


Circuit Diagram for Modulator



Circuit Diagram for DeModulator





THEORY:

Amplitude modulation can be produced by a circuit where the output is product of two input signals. Multiplication produces sum and difference frequencies and thus the side frequencies of the AM wave. Two general methods exist for achieving this multiplication, one involves a linear relation between voltage and current in a device and the second uses a linear device.A linear form of modulation of modulation causes a current I, of one frequency to pass through an impedance Z, whose magnitude varies at a second frequency. The voltage across this varying impedance is then given by

E = I sin1t * z sin2t

The above equation is the output is a result of multiplication of two frequencies. If one of them is carrier frequency and the other is the modulating frequency the result is an AM waveform.Modulated signal: The signal resulting from the process of modulation is referred to as modulated signal. Amplitude modulation is defined as a system of modulation in which the amplitude of the carrier is made proportional to the instantaneous amplitude of the modulating voltage . In AM , the amplitude of the carrier is varied by the modulating voltage. Whose frequency is lowerthan that of the carrier.

Let the carrier voltage Vc, the modulating voltage Vm represented as

Vc = Vc Sinwct

Vm = VmSinwmt

Modulation index : Vm/VC

Modulation index interms of Vmax and Vmin

M = (Vmax –Vmin)/(Vmax+Vmin)

Vc = (Vmax –Vmin)/2

Vm = (Vmax+Vmin)/2

Modulation index is lies between 0&1.

% of Modulation index = (Vm/VC)*100 (or) [(Vmax –Vmin)/(Vmax+Vmin)]*100

An AM wave described in its generated form as a function of time as follows:

S(t) = Ac[1+Ka*m(t) cos 2πfc t]

Modulus of (Ka*m(t)) is lessthan 1. It ensures that the function(1+ Ka*m(t)) is always positive. The carrier frequency is much greater than the modulating frequency.

In a communication system a high frequency carrier is modulated by the low frequency signal. The modulated carrier is transmitted by the transmitter antenna. At the receiver we have to recover the information back from the modulated carrier. The process of separation of signal from the carrier is called demodulation or detection. the demodulation circuit diagram is a linear diode detector. In this circuit the linear portion of dynamic characteristics of diode is used and hence the circuit is a linear detector. It consists of a half wave rectifier followed by a capacitor input filter. Input to the circuit is an AM wave with a high frequency carrier and a low frequency envelope corresponding to the signal. The diode cuts-off the negative going portion of the AM wave. Capacitor „C‟ charges up to the peak of the carrier cycle through the low resistance rd and then during negative half cycle tries to discharge through relatively high resistance RL. Capacitor value is so chosen that this discharge is very small in time between carrier half cycles. Hence the capacitor voltage tends to follow the envelope of the carrier and the voltage available across RL is simply the modulation envelope

superimposed on a constant level. A dc level in the output comes because the current through diode flows in the form of pulses occurring at the peak of each carrier cycle.
When the input to detector circuit is a AM waveform then the one of the component in VR cannot be assumed to be constant all the time. Actually it is constant over a few cycles of carrier in which time it is assumed that modulating signal being low frequency would not have changed appreciably. Due to this reason the measurement of detection efficiency can be done on an un modulated carrier because VR would be expected to be constant

PROCEDURE:-

1. Measure the frequency & amplitude (p-p) of the fixed carrier signal present on the kit.

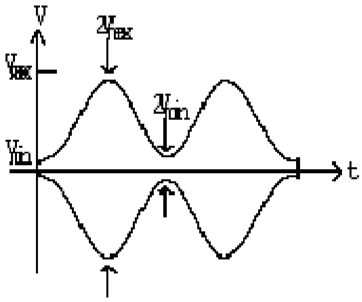
2. Connect the circuit as per the given circuit diagram.

3. Apply fixed frequency carrier signal to carrier input terminals.

4. Apply modulating signal from function generator of 1VP-P of 500Hz.

5. Note down and trace the modulated signal envelop on the CRO screen.

6. Find the modulation index by measuring Vmax and Vmin from the modulated (detected/ traced) envelope.
M=(Vmax –Vmin)/(Vmax+Vmin)



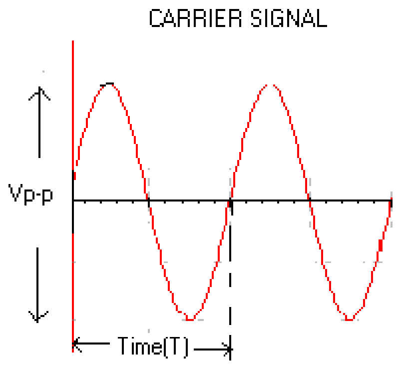
7. Repeat the steps 5 & 6 by changing the frequency or/& amplitude of the modulating signal so as to observe over modulation, under modulating and perfect modulation.

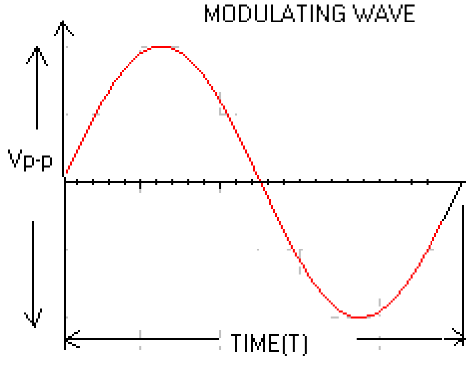
8. For demodulation, apply the modulated signal (A.M) as an input to the demodulator and verify the demodulated output with respect to the applied modulating signals and their respective outputs.

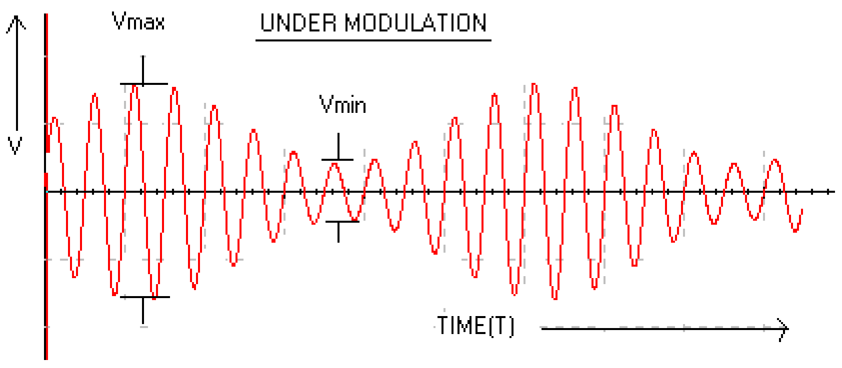
TABULAR COLUMN:

Modulator	V_{max}	V_{min}	$M=(V_{max}-V_{min})/(V_{max}-V_{min})$
100%			
Over			
Under			

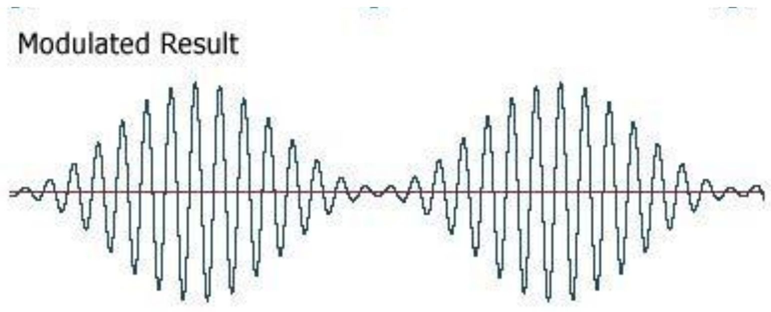
EXPECTED WAVEFORMS:

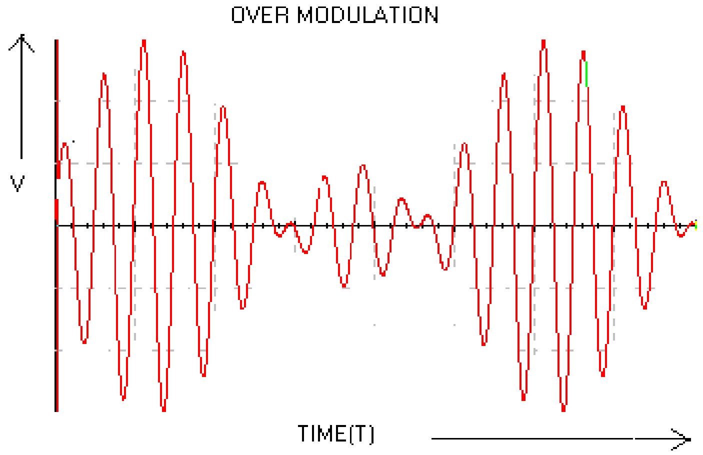




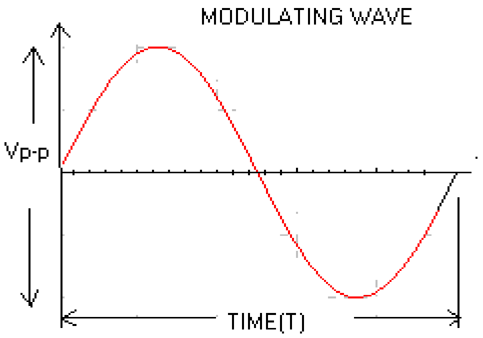


PERFECT MODULATION





Demodulated signal



RESULT:

SIGNATURE OF THE INCHARGE

QUESTIONS

1.Define AM and draw its spectrum?
2.Draw the phases representation of an amplitude modulated wave?
3.Give the significance of modulation index?
4.What are the different degree of modulation?
5.What are the limitations of square law modulator?
6.Compare linear and nonlinear modulators?
7.Compare base modulation and emitter modulation?
8.Explain how AM wave is detected?

MATLAB CODE

```
fs=8000;
fm=20;
fc=500;
Am=1;
Ac=1;
t=[0:0.1*fs]/fs;
m=Am*cos(2*pi*fm*t);
c=Ac*cos(2*pi*fc*t);
ka=0.5;
u=ka*Am;
s1=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);
subplot(4,3,1:3);
plot(t,m);
title('Modulating or Message signal(fm=20Hz)');
subplot(4,3,4:6);
plot(t,c);
title('Carrier signal(fc=500Hz)');
subplot(4,3,7);
plot(t,s1);
title('Under Modulated signal(ka.Am=0.5)');
Am=2;
ka=0.5;
u=ka*Am;
s2=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);

subplot(4,3,8);
plot(t,s2);
title('Exact Modulated signal(ka.Am=1)');
Am=5;
ka=0.5;
u=ka*Am;
s3=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);
subplot(4,3,9);
plot(t,s3);
title('Over Modulated signal(ka.Am=2.5)');
r1= s1.*c;
[b a] = butter(1,0.01);
mr1= filter(b,a,r1);
subplot(4,3,10);
plot(t,mr1);
r2= s2.*c;
[b a] = butter(1,0.01);
mr2= filter(b,a,r2);
subplot(4,3,11);
plot(t,mr2);
r3= s3.*c;
```

```
[b a] = butter(1,0.01);
mr3= filter(b,a,r3);
subplot(4,3,12);
plot(t,mr3);
```

SIMULATED OUTPUT WAVE FORMS FOR AM

SIGNATURE OF THE LAB INCHARGE

ATTACH GRAPH SHEET HERE

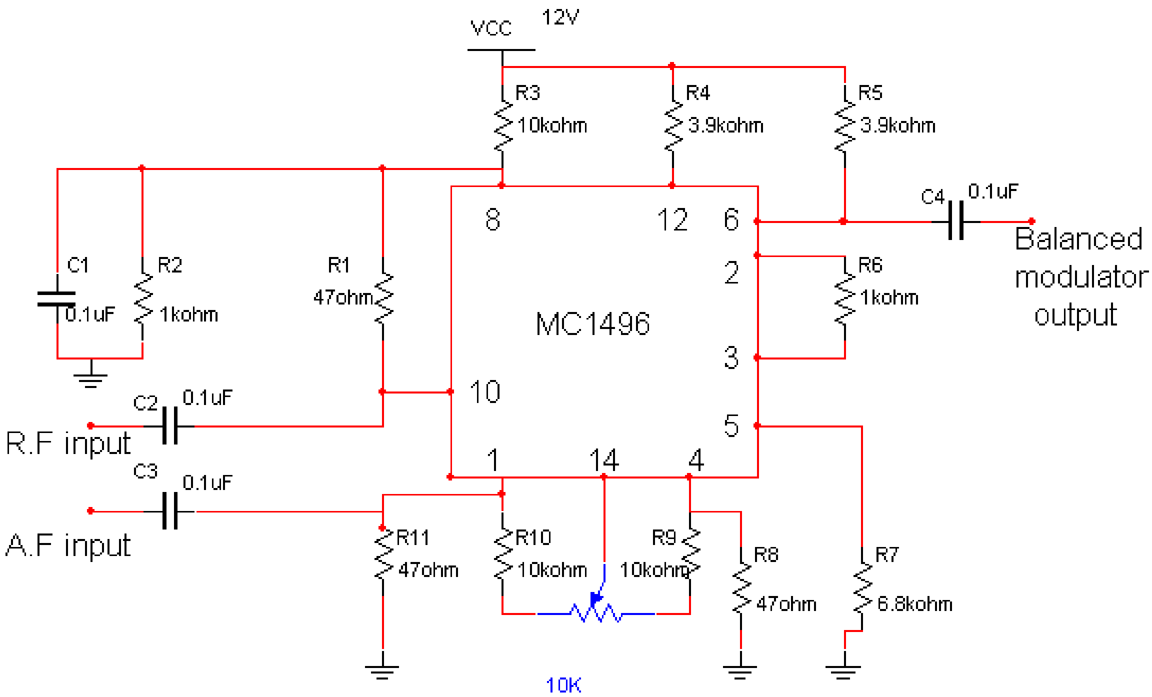
EXPERIMENT NO-2

BALANCED MODULATOR

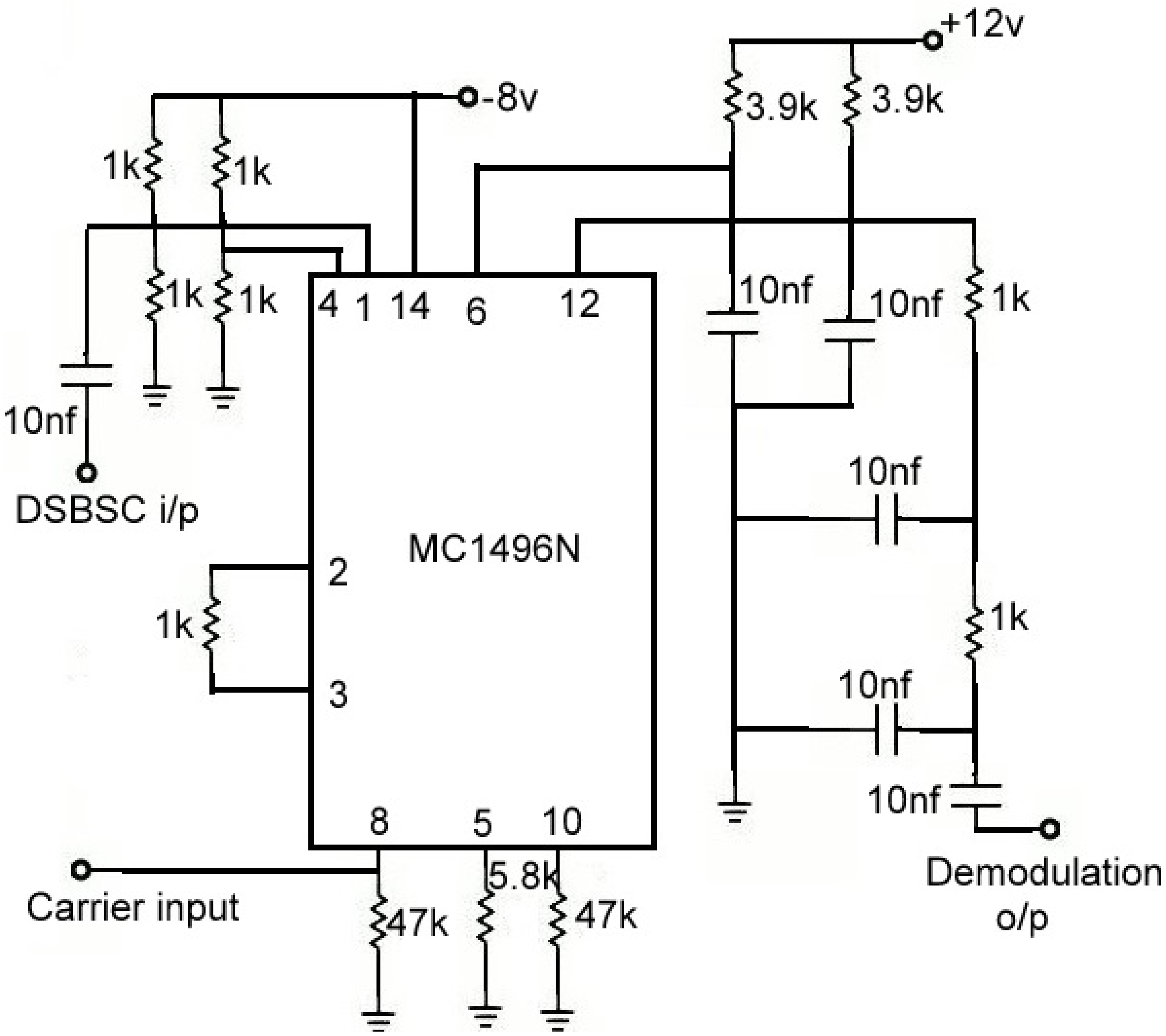
AIM: To study the following of the Balanced Modulator as a DSB-SC Generator.

APPARATUS:
1. Balanced modulator trainer kit
2. C.R.O (20MHz)
3. Connecting cords and probes
4. Function generator (1MHz)

CIRCUIT DIAGRAM OF MODULATOR:



CIRCUIT DIAGRAM OF DEMODULATOR:



THEORY:

The carrier of amplitude modulation wave does not convey any information. It is obvious from the fact that the carrier component remains constant in amplitude and frequency. No matter what the modulating signal does. It is thus, seen that no information is conveyed by the carrier. If the carrier is suppressed, only the side bands remains and a saving of two third powers can achieve at 100% modulation such suppression of carrier doesn"t affect the massage signal in any way. This idea has resulted in the evolution of suppressed carrier modulation. Thus, the short coming of the conventional AM in regard of power wastage is overcome by suppressing the carrier from the modulated wave resulting in double side band suppressed carrier modulation. A balanced is used to generate DSBSC wave. A DSBSC signal is basically the product of the base band signal and the carrier wave.

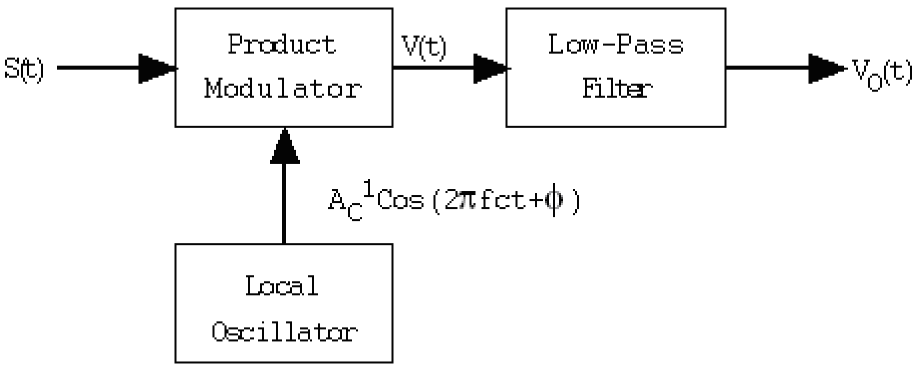
S(t) = m(t) *c(t)

Where m(t) is base band signal , C(t) is carrier signal and C(t) = Ac cost 2Π fct

The modulated wave under goes a phase reversal when ever base band signal m(t) crosses zero.Spectrum of base band signal.The fourier transform of the DSBSC modulated signal is
S(f) = AC/2 [(M(f-fc) + M(f+fc)]
Where M (f) is the fourier trans form of m(t)

And fc is frequency of the carrier.
The band width of DSBSC signal is same as that of conventional AM i.e., 2W.
The base band signal m(t) can be uniquely recovered from a DSB-SC wave S(t) by first multiplying s(t) with a locally generated sinusoidal wave and then low-pass filtering the product, as in fig. below. It is assumed that the local oscillator signal is exactly coherent or synchronized, in both frequency and phase, with the carrier wave C(t) used in the product modulator to generate S(t). This method of demodulation is known as **Coherent or Synchronous**

DEMODULATION.



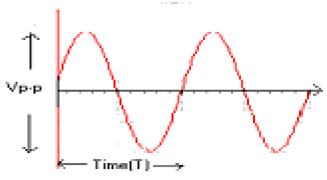
PROCEDURE:-

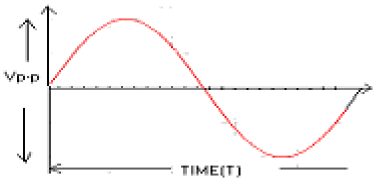
1.DSB-SC GENERATION

1)For the same circuit apply the modulating signal(AF) frequency in between 1Khz to 5Khz having 0.4 VP-P and a carrier signal(RF) of 100KHz having a 0.1 VP-P .
2) Observe the output of the kit on CRO and adjust the carrier null potentiometer until the DSB-SC modulated wave is obtained

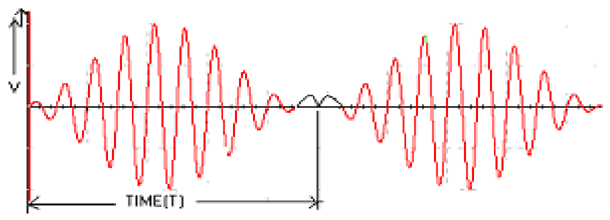
EXPECTED WAVE FORMS:

CARRIER SIGNAL (RF) MODULATING SIGNAL (AF)





DSC-SC MODULATED OUTPUT

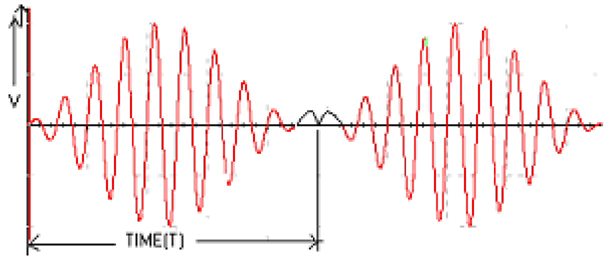


DSB-SC Demodulation:

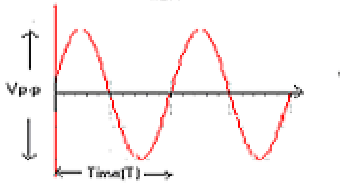
1. Apply DSB-SC signal as an input the synchronous detector and RF generator output(or carrier) to RF input of synchronous detector.
2. Observe the synchronous detector output on CRO and compare it with the original AF signal.

EXPECTED WAVE FORMS:

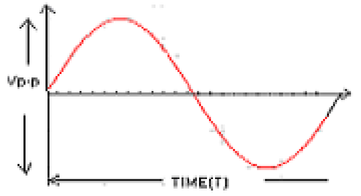
DSC-SC INPUT SIGNAL



CARRIER INPUT SIGNAL (RF)



DEMODULATED OUTPUT SIGNAL (AF)



RESULT:

SIGNATURE OF THE LAB INCHARGE

QUESTIONS

1.What are the two ways of generating DSB_SC.
2.What are the applications of balanced modulator?
3.What is the advantages of suppressing the carrier?
4.What are the advantages of balanced modulator?
5.What are the advantages of Ring modulator?

MATLAB CODE:

```
t =0:0.000001:.001;
Vm= 1;
Vc= 1;
fm = 2000;
fc= 50000;
m_t = Vm*sin(2*pi*fm*t);
subplot(4,1,1);
plot(t,m_t);
c_t = Vc*sin(2*pi*fc*t);
subplot(4,1,2);
plot(t,c_t);
subplot(4,1,3);
s_t = m_t.*c_t;
hold on;
plot(t,s_t);
plot(t,m_t,'r:');
plot(t,-m_t,'r:');
hold off;
r = s_t.*c_t;
[b a] = butter(1,0.01);
mr= filter(b,a,r);
subplot(4,1,4);
plot(t,mr);
```

SIMULATED OUTPUT WAVEFORMS:

SIGNATURE OF THE LAB INCHARGE

ATTACH GRAPH SHEET HERE

EXPERIMENT NO-3
SSB MODULATOR AND DETECTOR (PHASE SHIFT METHOD

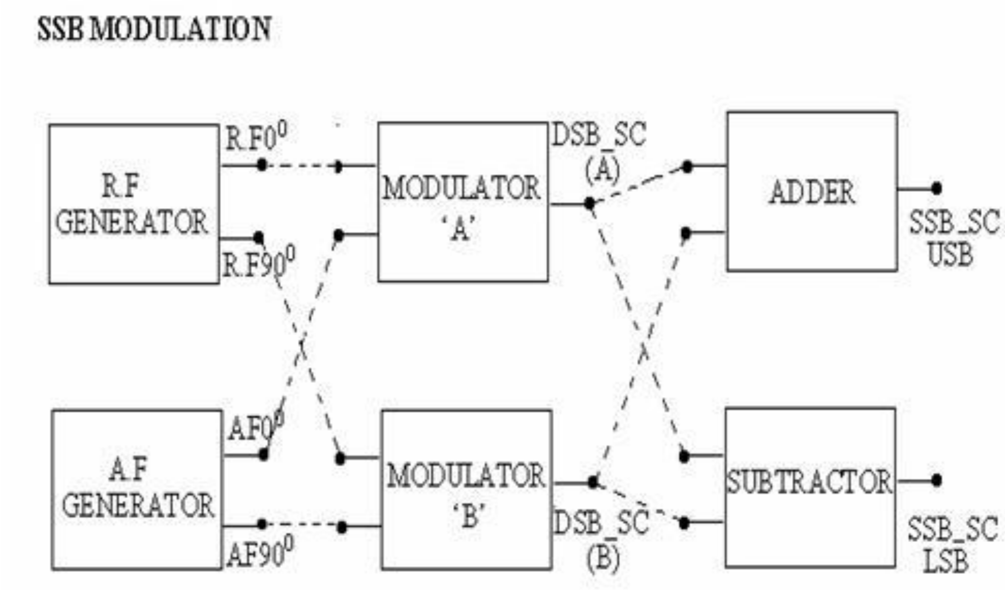
AIM : To generate SSB using phase method and demodulation of SSB signal using Synchronous detector.

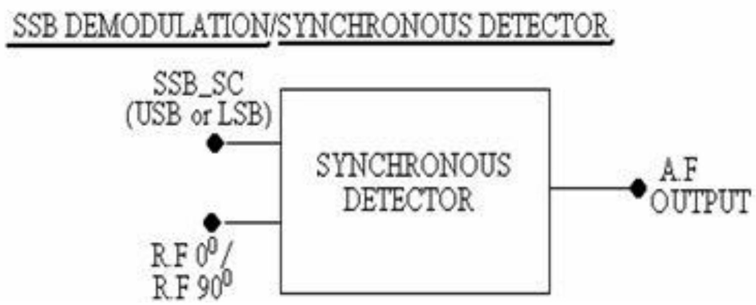
APPARATUS:

1.SSBtrainer kit

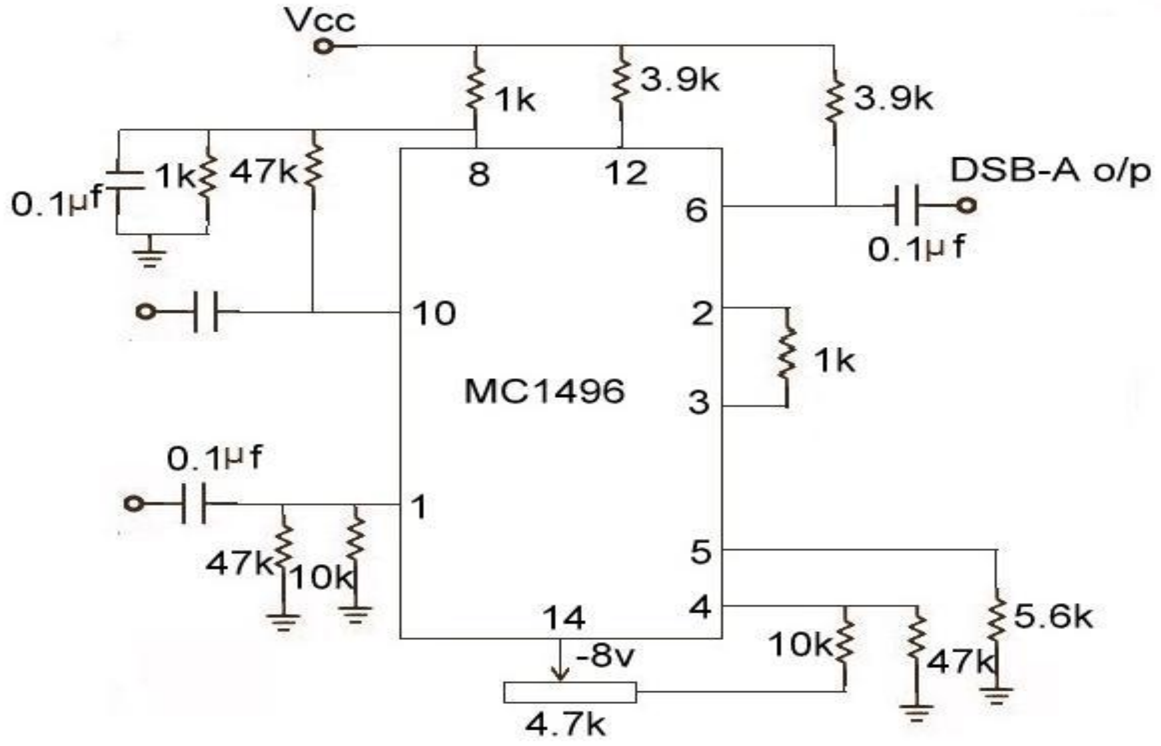
2. C.R.O (20MHz)

BLOCK DIAGRAM

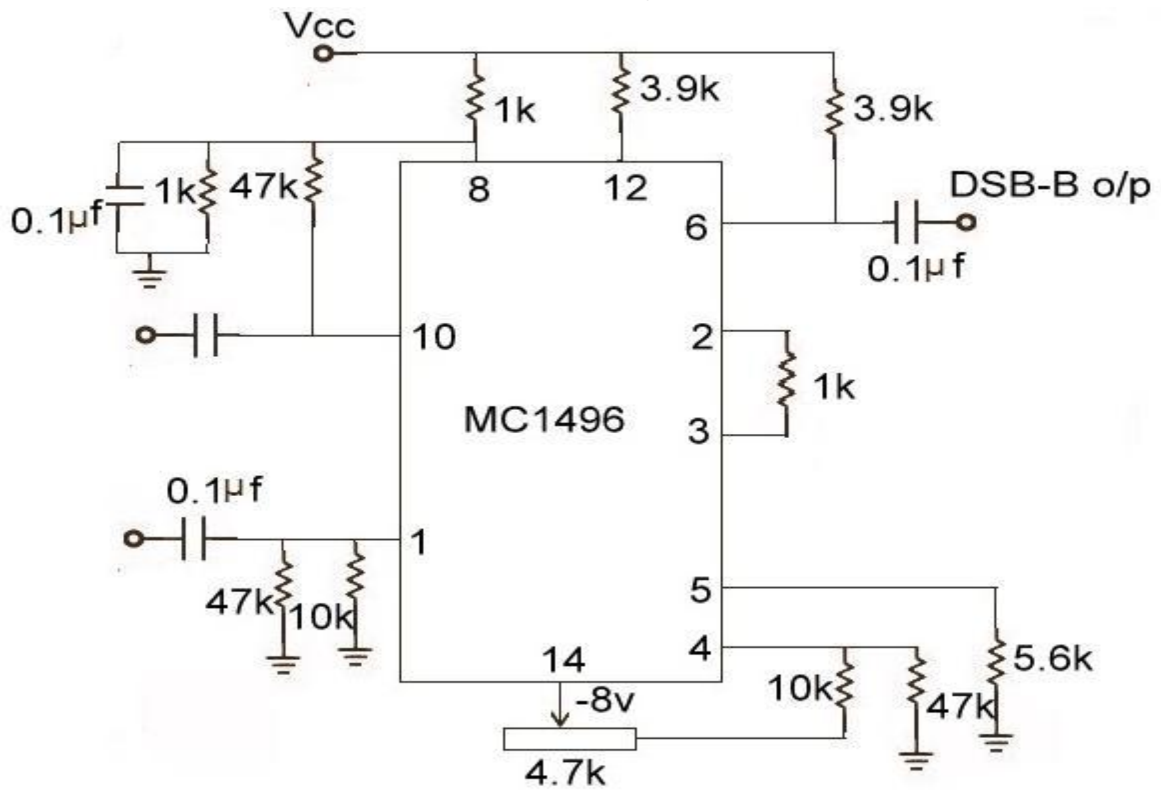




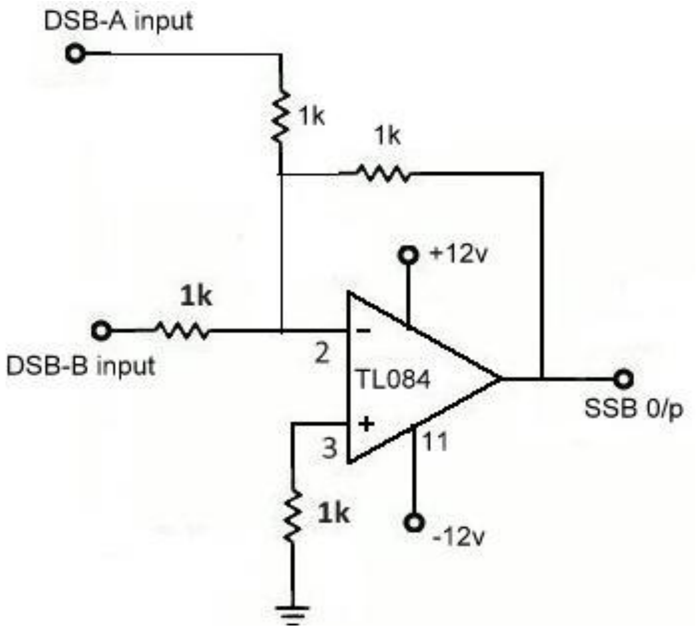
INTERNAL DIAGRAMS OF SSB TRAINER KITCIRCUIT DIAGRAM:



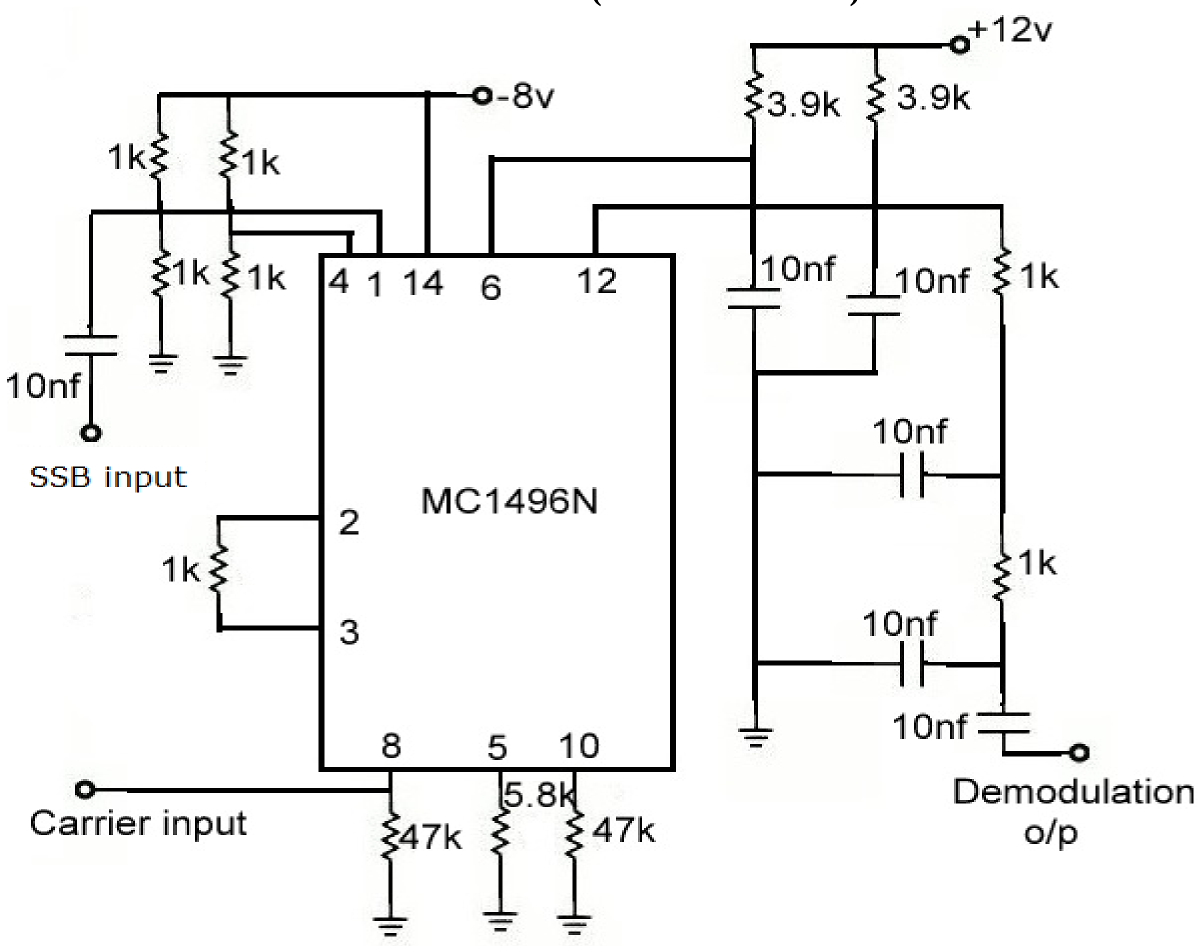
DSB-SC A Modulator



DSB-SC B Modulator



SSB—SUMMER (SUBSTRACTOR)



DEMODULATOR

THEORY:

Single side band signal generation using Phase shift method and demodulation of SSB signal using Synchronous detector. This exp consists of

1.R.F generator.
2.A.F generator.
3.Two balanced modulators.
4.Synchronous detector
5.Summer
6.Subtractor

These circuits are simple summing and subtracting amplifiers using OP-AMP. IC TL084 is used as an active component, TL 084 is a FET input general purpose quad OP-AMP integrated circuit.

The phase shift method makes use of two balanced modulators and two phase shift networks as shown in figure. One of the modulators receives the carrier signal shifted by 900 and the modulating signal with 00 (sine) phase shift, where as the other receives modulating signal shifted by 900 (cosine) and the carrier (RF) signal with 00 phase shift voltage.

Both modulators produce an output consisting only of sidebands. It will be shown that both upper sidebands leads the reference voltage by 900, and the other lags it by 900. The two lower side bands are thus out of phase and when combined in the adder, they cancel each other. The upper side bands are in phase at the adder and therefore they add together and gives SSB upper side band signal. When they combined in the subtractor, the upper side bands are canceling because in phase and lower side bands add together and gives SSB lower side band signal.

PROCEDURE:

SSB MODULATION

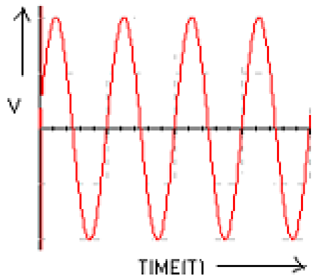
1.Connect the circuit as per the given circuit diagram.
2.Switch on the kit and measure the output of regulated power supplies positive and negative voltages.
3.Observe the outputs of RF generators using CRO .Where one output is 00phase the another is 900 phase shifted(or) is a sine wave and shifted w.r.t other (or) is a cosine wave.
4. Adjust the RF output frequency as 100KHz and amplitude as 0.2 Vp-p (Potentiometers are provided to vary the output amplitude & frequency).
5. Observe the two outputs of AF generator using CRO.
6. Select the required frequency (2kHz, 4kHz, 6kHz) form the switch positions for A.F.
7. Adjust the gain of the oscillator by varying the AGC potentiometer and keep the amplitude of 10Vp-p.
8. Measure and record the above seen signals & their frequencies on CRO.
9. Set the amplitude of R.F signal to 0. 2Vp-p and A.F signal amplitude to 8Vp-p and connect AF-0^0 and RF-90^0 to inputs of balanced modulator A and observe DSB-SC(A) output on CRO. Connect AF-900 and RF-00 to inputs of balanced modulator B and observe the DSB-SC (B)out put on CRO and plot the same on graph.
10. To get SSB lower side band signal connect balanced modulator outputs (DSB-SC) to subtractor and observe the output wave form on CRO and plot the same on

graph.
11. To get SSB upper side band signal, connect the output of balanced modulator outputs to summer circuit and observe the output waveform on CRO and plot the same on graph.
12.Calculate theoretical frequency of SSB (LSB & USB) and compare it with practical value.
USB = RF frequency + AF frequency. (Fc+Fm)
LSB = RF frequency – AF frequency. (Fc-Fm)

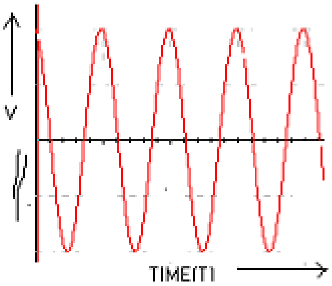
SSB DEMODULATION :
1. Connect the SSB signal from the summer or subtractor at SSB signal input terminal of synchronous detector.
2. Connect RF signal (00) at RF input terminal of the synchronous detector.
Observe the detector output on CRO and compare it with the modulating signal(AF Signal) and plot the same on graph.

EXPECTED WAVE FORMS: -

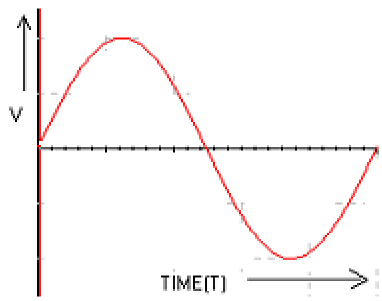
RF CARRIER SIGNAL(0^0)



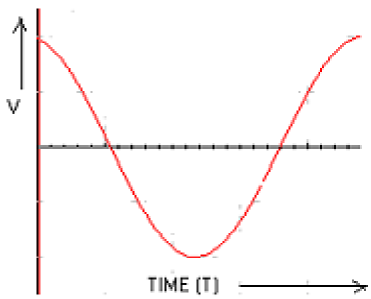
RF CARRIER SIGNAL(90^0)



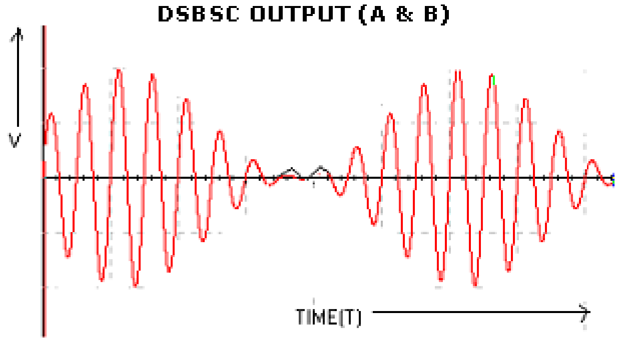
AF MODULATING SIGNAL(0^0)

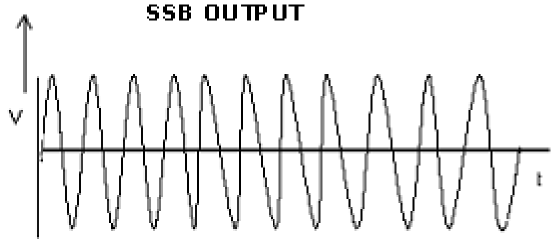


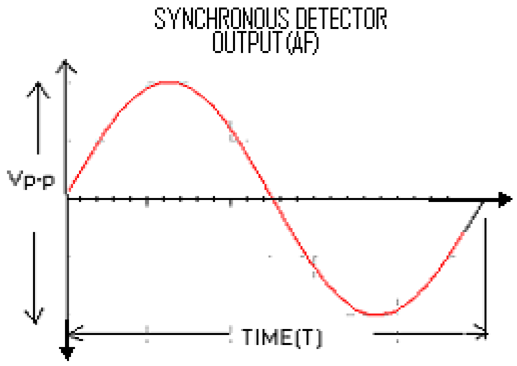
AF MODULATING SIGNAL(90^0)



DSBSC OUTPUT (A & B)







RESULT:.

SIGNATURE OF THE LAB INCHARGE

QUESTIONS

1.What are the two ways of generation of SSB wave?
2.What are the features of filter method generation of SSB?
3.What are the adventages of phase shift method of SSB generation?
4.What are the disadvantages of phase shift method of SSB generation?
5.What are the advantages of SSB-SC AM?
6. What are the disadvantages of SSB-SC AM?
7. What are the applications of SSB-SC AM?

MATLAB CODE:

```
s=8000;
fm=20;
fc=50;
Am=1;
Ac=1;
t=[0:.1*fs]/fs;
subplot(4,2,1);
m1=Am*cos(2*pi*fm*t);
plot(t,m1);
title('Message Signal');
m2=Am*sin(2*pi*fm*t);
subplot(4,2,2)
c1=Ac*cos(2*pi*fc*t);
plot(t,c1)
title('Carrier Signal');
c2=Ac*sin(2*pi*fc*t);
subplot(4,2,3)
% Susb=0.5* Am*cos(2*pi*fm*t).* Ac*cos(2*pi*fc*t) -- 0.5* Am*sin(2*pi*fm*t).*
Ac*sin(2*pi*fc*t);
Susb=0.5*m1.*c1-0.5*m2.*c2;
plot(t,Susb);
title('SSB-SC Signal with USB');
subplot(4,2,4);
Slsb=0.5*m1.*c1+0.5*m2.*c2;
plot(t,Slsb);
title('SSB-SC Signal with LSB');
r = Susb.*c1;
[b a] = butter(1,0.0001);
mr= filter(b,a,r);
subplot(4,2,5);
plot(t,mr);
```

SIMULATED OUTPUT

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EXPERIMENT NO-4

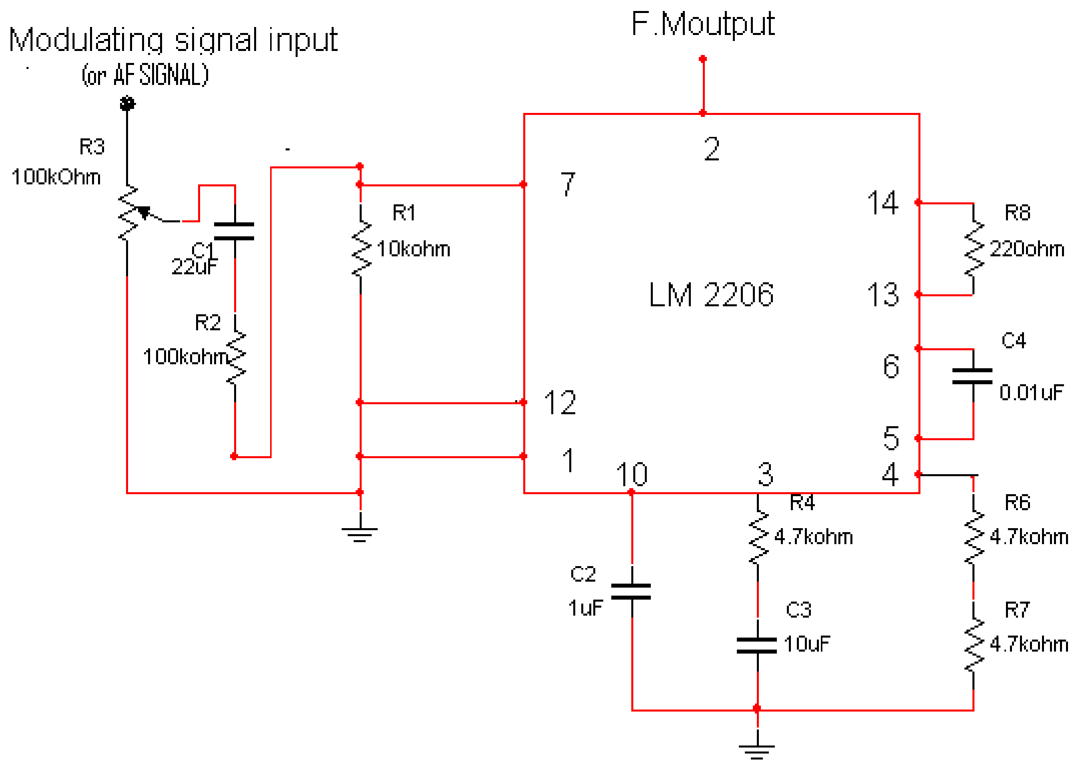
FREQUENCY MODULATION & DEMODULATION

AIM: To study the functioning of frequency modulation & demodulation and to calculate the modulation index.

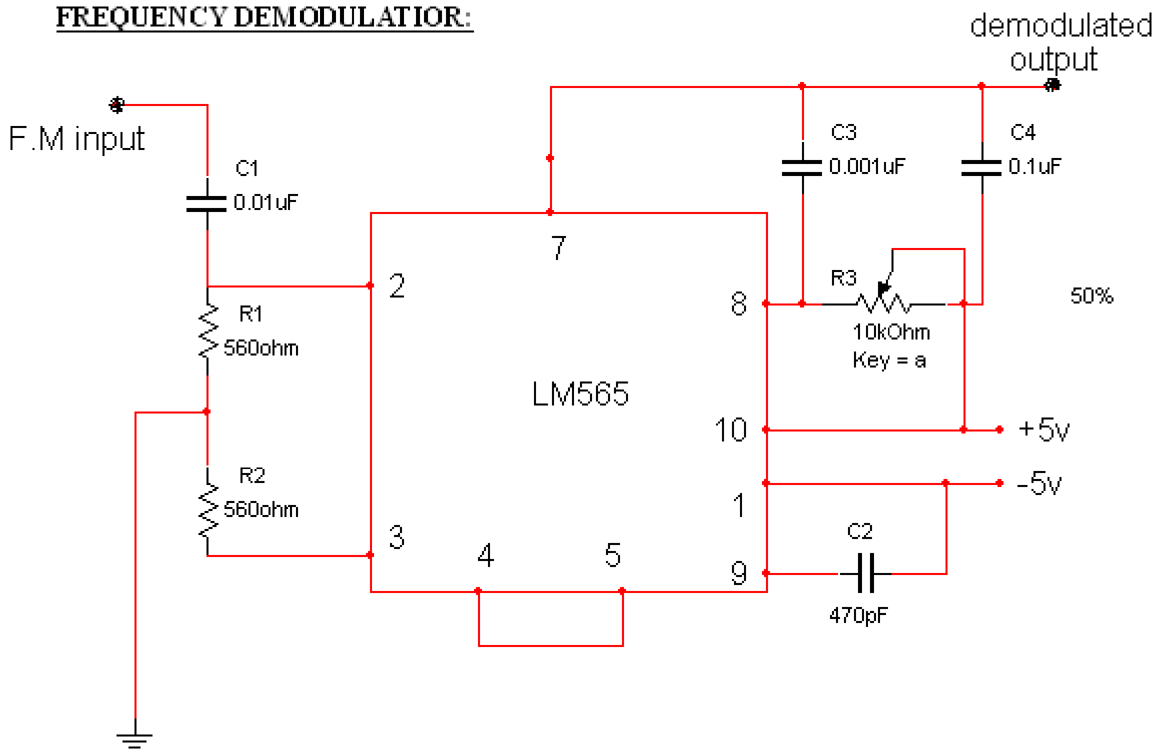
APPARATUS:
1. Frequency modulation & demodulation trainer kit.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Connecting chords & probes.

CIRCUIT DIAGRAM:

FREQUENCY MODULATIOR



FREQUENCY DEMODULATIOR:



THEORY:
This kit consists of wired circuitry of:

1. AF generator.
2. Regulated power supply
3. Modulator.
4. Demodulator.

Modulator:
This has been developed using XR-2206 integrated circuit. The IC XR-2206 is a monolithic Function generator; the output waveforms can be both amplitude and frequency modulated by an external voltage. Frequency of operation can be selected
externally over a range of 0.01 MHz. The circuit is ideally suited for communications, instrumentations and function generator applications requiring sinusoidal tone, AM, FM or FSK generation. In this experiment, IC XC-2206 is connected to generate sine wave, which is used as a carrier signal. The amplitude of carrier signal is 5vPP of 100 KHz frequencies.

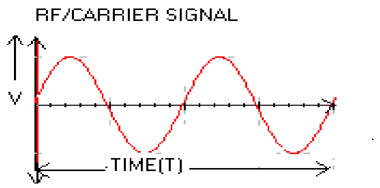
Demodulator:
This had been developed using LM4565 integrated circuit. The IC LM565 is a general-purpose phase locked loop containing a stable, highly linear voltage controlled oscillator for low distortion FM demodulation.
The VCO free running frequency f0 is adjusted to the center frequency of input frequency modulated signal i.e. carrier frequency which is of 100 KHz. When FM signal is connected to the demodulator input, the deviation in the input signal (FM signal) frequency which creates a DC error voltage at output of the phase comparator which is proportional to the change of

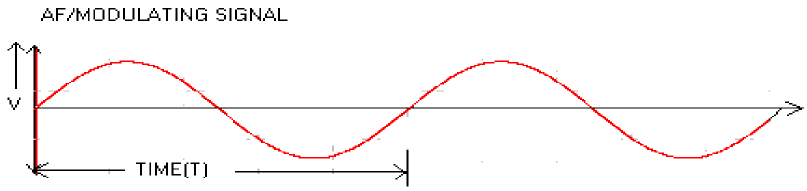
frequency f. This error voltage pulls the VCO to the new point. This error voltage will be the demodulated version of the frequency modulated input signal.

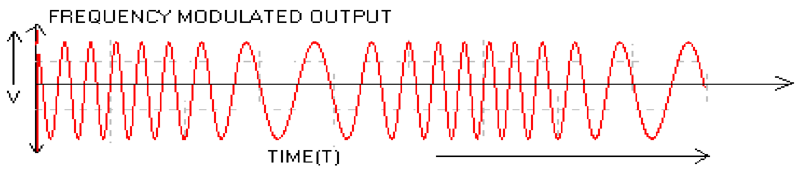
PROCEDURE:

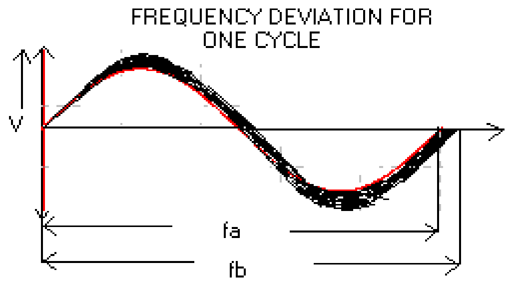
1. Switch on the power supply of the kit (without making any connections).
2. Measure the frequency of the carrier signal at the FM output terminal with input terminals open and plot the same on graph.
3. Connect the circuit as per the given circuit diagram.
4. Apply the modulating signal of 500HZ with 1Vp-p.
5. Trace the modulated wave on the C.R.O & plot the same on graph.
6. Find the modulation index by measuring minimum and maximum frequency deviations from the carrier frequency using the CRO.
7. Repeat the steps 5& 6 by changing the amplitude and /or frequency of the modulating Signal.
8. For demodulation apply the modulated signal as an input to demodulator circuit and compare the demodulated signal with the input modulating signal & also draw the same on the graph.

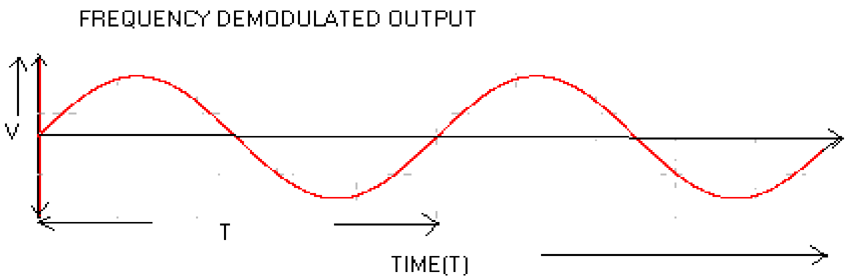
EXPECTED WAVEFORMS











NOTE: Note down all the input and output wave forms of the signals applied and obtained respectively.

RESULT:

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QUESTIONS

1.Define frequency modulation?
2.Mention the advantages of indirect method of FM generation?
3.Define modulation index and frequency deviation of FM?
4.What are the advantages of FM?
5.What is narrow band FM?
6.Compare narrow band FM and wide band FM?
7.Differrntiate FM and AM?
8.How FM wave can be converted into PM wave?
9,State the principle of reactance tube modulator?
10.Draw the circuit of varactor diode modulator?
11.What is the bandwidth of FM system?
12.Want is the function of FM discriminator?

MATLAB CODE:

```
%The frequency modulation(FM)waveform in time and frequency domain.
%fm=35HZ,fc=500HZ,Am=1V,Ac=1V,B=10
function fmdm
fs=10000;
Ac=1;
Am=1;
fm=35;
fc=500;
B=10;
t=(0:.1*fs)/fs;
wc=2*pi*fc;
wm=2*pi*fm;
m_t=Am*cos(wm*t);
subplot(5,1,1);
plot(t,m_t);
title('Modulating or Message signal(fm=35Hz)');
c_t=Ac*cos(wc*t);
subplot(5,1,2);
plot(t,c_t);
title('Carrier signal(fm=500Hz)');
s_t=Ac*cos((wc*t)+B*sin(wm*t));
subplot(5,1,3);
plot(t,s_t);
title('Modulated signal');
d=demod(s_t,fc,fs,'fm');
subplot(5,1,4);
plot(t,d);
title('demodulated signal');
```

STIMULATED OUTPUT WAVEFORM:

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EXPERIMENT NO-5

STUDY OF SPECTRUM ANALYZER AND ANALYSIS OF AM & FM SIGNALS

1 GHz Spectrum Analyzer INSA10

Frequency Range : 0.15MHz to 1050MHsz
4 Digit Display
-100 to + 13 dBm Amplitude range
Tracking Generator Frequency range : 0.15 MHz to 1050 MHz
Output Voltage +1 dBm to dBm
The new Intrx INSA10, 1 GHz Spectrum Analyzer extends it‟s application beyond 1GH, Both fine and coarse center frequency controls, combined with a scan provide simple frequency domain measurements from 100kHz/div to 100MHz/div to 100MHz. The numeric 4½ digit LED readout is provide d, to display either center or marker frequency. INSA10 includes tracking generator. The instrument is suitable for pre-compliance testing during development prior to third party testing. A near field sniffer probe set, (optional) can be used to locate cable and PC board emission of hot spots and evaluate EMC problems at the breadboard and prototype level.
Application of Spectrum Analyzer:
1.Testing and fault analysis in cable and wireless systems including remote controls, cordless phones cable TV, communication equipments , as well as good comparison and analysis to frequency of signals.
2.Testing and fault finding in mobile phones, RF circuits baseband signals local oscillator of RF circuits.
3.Electromagnetic Compatible (EMC) testing , to test and locate disturbing nearby frequency from other sources in parent instrument. It can output AM/FM demodulated signal from phone jack, to identify the disturbing broadcast station.
4.Widely used in production development education labs research laboratories.
Technical Specifications
Frequency
Frequency Range : 0.15 MHz to 1050 MHz
Center Frequency Display Accuracy : +100kHz
Marker Accuracy : ± (0.1% span +100kHz)
Frequency Display Resolution :100kHz (4 ½ -digit)
Frequency Scan width :100kHz/div to 100 MHz /div in 1-2-5 steps and 0 Hz/div (Zero scan)
Frequency Scan width accuracy: ±10%
Frequency Stability: better than 150Khz/hour
IF Bandwidth(-3dB): Resolution 400kHz , 20kHz
Video Filter ON: 4kHz
Sweep Rate :43Hz

Amplitude
Measurement Range : -100 dB to + 13dBm
Screen Display Range: 80 Db (10 Db/div)

Reference level : -27 dBm to +13 dBm (in 10 dB steps)
Reference level Accuracy: ±2 dB
Average Noise Level: -99 dBm (20 kHz BW)
Distortion :< -55 dB; 2nd and 3rd harmonic 3rd order intermod.: -70 dBc(two signals >3MHz apart)
Sensitivity:< 5 dB above average noise level
Log Scale Fidelity: 2± dB(without attn.) Ref: 55 MHz input
Input Impedance: 50Ω
Input Connector : BNC
Input Attenuator: 0 to 40 Db (4x10 dB steps)
Input Attenuator Accuracy : ± 1dB /10 dB step
Max.Input Level: +10 dBm +25 Vdc (0 dB attenuation), + 20 dBm (40 dB attennuation)

Tracking Generator

Output Level Range : -50 dB to +1 dBm (in 10 dB steps and var.)\
Output Attenuator Accuracy: ± 1 dB
Output Impedance : 50Ω (BNC)
Frequency Range : 0.15 MHz to 1050MHz
Frequency Response: ± 1.5 dB
Radio Frequency Interference(RFI): < 20dbc

General Information

AM/FM Demodulation output for head- sets Permissible load > 8Ω
Display : CRT 6 in, 8x10 div, internal graticule,
Trace Rotation: Adjustable on front panel
Supply : 230 , ± 10% 50Hz , AC
Power Consumption : 20 W approx.
Operating Conditions: 0 ~ 40 C RH 95%
Protective System : Safety Class I (IEC 1010-1)
Dimension: W285 , H 125, D380 mm
Weight: 7 kg approx.
Front Panel Controls

1. **Focus :** Beam sharpness adjustment.

2. **Intens :** Beam intensity adjustment.

3. **Power :** If power is switched to ON position, a beam will be visible on the screen after approximately 10 sec.

4. **TR(Trace Rotation):** In spite of Mumetal- shielding of the CRT, effects of the earth‟s magnetic field on horizontal trace position cannot be completely avoided. A potentiometer accessible through an opening can be used for correction. Slight Pincushion distortion is unavoidable and cannot be corrected

5. **Marker ON/OFF :** When the MARKER pushbutton is set to the OFF position the CF indicator is field on the horizontal. When the switch is in the ON position, MK is lit and the display shows the center frequency. The marker is shown on the screen as sharp peak. The marker frequency is adjustable by means of the marker knob and can be aligned with a spectral line. Switch off the marker before taking correct amplitude readings.

6. **CF/MK :** The CF LED is lit when the digital display shows the center frequency. The center frequency is the frequency which is displayed in the horizontal center of the CRT. The MK LED is lit when the marker pushbutton is in the ON position. The digital display shows the marker frequency in that case.

7. **Digital Display :** Frequency / Marker Frequency) 7-segment. Display with 100kHz resolution.

8. **Uncal :**Blinking to this LED indicates incorrectly displayed amplitude values. This is due to scanwidth and filter setting combination switch give to low amplitude readings because the If- filters have not being settled. This may occur when the scanned frequency range (SCAWIDTH) js too large compared to the IF bandwidth (20kHz), and/or the video filter bandwidth (4kHz). Measurements in the case can either be taken without a video filter or the scanwidth in this case can either be taken without a video filter or the scanwidth has to be decreased.

9. **Center frequency – Coarse/Fine :** Bothe rotary knobs are used for center frequency setting. The center frequency is displayed at the horizontal center of the screen.

10. **Bandwidth :** Selects between 400 kHz and 20kHz IF bandwidth. If a bandwidth of 20kHz is selected, the noise level decreases and the selectivity is improved. Spectral lines which are relatively close together can be distinguished. As the small signal transient response requires a longer time this causes incorrect amplitude values if the scanwidth is set at too wide a frequency span. The UNCAL. LED will indicate this condition.

11. **Video Filter:** Video filter may be used to reduce noise on the screen. It would be within or just above the medium noise level. The filter bandwidth is 4kHz.

12. **V-Position :** Control for adjusting the vertical beam position.

13. **Input:** The BNC 50 Q input of the Spectrum Analyzer. Without input attenuation the maximum permissible input voltages of ±25V DC and +10dbm AC must not be exceeded. With the maximum input attenuation of 40dB the maximum input voltage is +20 dBm. The maximum dynamic range of the instrument is 70dB. Higher input voltages exceeding the reference level cause signal compression and inter modulation. Those effects will lead to erroneous displays. If the input level exceeds the reference level. The input level attenuation must be increased.

14. **Attenuator :** The input Attenuator consists of four 10db attenuators reducing the signal height before entering the 1st mixer. Each attenuator is active if the push button is depressed. The correlation of selected attenuation. Reference level, and baseline level(noise level) is according to the following listing.
The reference level is represented by the upper horizontal graticule line. The lowest horizontal graticule line indicates the baseline. The vertical graticule is subdivided in 10db steps. As previously pointed out the maximum permissible input voltages may not be exceed 3dbm. This is extremely important because it is possible that the Spectrum Analyzer will only show a partial spectrum of currently applied signals. Consequently input signals might be

applied with excessive levels outside the displayed frequency range leading to the destruction of the input attenuator and /or the 1st mixing stage. Also refer to INPUT. The highest attenuation (4×10dB) and the highest usable frequency range (scanwidth setting 50 MHz/div) should be selected prior to any spectral to the INSA10 input. This permits the detection of any spectral lines which are within the maximium measurable and displayable frequency range if the center frequency is set to 500MHz . If the baseline tends to move upwards when the attenuation is decreased it may indicate spectral lines outside the maximum displayable frequency range(i.e. 1200MHz) with excessive amplitude.

15. **Scanwidth :** The SCANWIDTH selectors allow to control the scanwidth per division of the horizontal axis. The frequency/div. can be increased by means of the >button, and decreased by means of the button Switching is accomplished in 1-2-5 steps from 100 kHz/div to 100kHz/div. The width of the scan range is displayed in MHz/div.and refers to each horizontal division on the graticule. The center frequency is indicated by the vertical graticule line at middle of the horizontal axis. If the center frequency and the scanwidth settings are correct the X axis has a length of 10 divisions On scanwidth settings lower than 100MHz only a part of the entire frequency range is displayed. When SCANWIDTH is set to 100MHz/div. and if center frequency is set to 500MHz. the displayed frequency range extends to the right by 100MHz per division ending at 1000MHz [500MHz +(5×100MHz)]. The frequency decreases to the left in a similar way. In this case the left graticule line corresponds to 0 Hz. Whth these settings, a spectral line is visible which is referred to as "Zero Frequecy". It is the 1st LO(oscillator) which becomes visible when its frequency passes the first IF filter, This occurs when the center frequency is low relative to the scanwidth range selected. The "Zero Frequency" is different in level in every instrument and therefore cannot be used as a reference level. Spectral lines displayed left of the "Zero Frequency Point" are so-called image frequencies. In the ZERO SCAN mode the Spectrum Analyzer operates like Spectral line(s) passing the IF filter cause a level display (selective voltmeter function). The selected scanwidth/div. settings are indicated by a number of LEDs above the range setting push buttons.

16. **Attenuation :** Output level attenuator with four 10dB attenuators which allows the signal to be reduced prior to reaching the OUTPUT socket. All four attenuators are equal and can be activated by pressing the respective push button. Therefore, it is irrelevant which attenuators are used to reach e.g. a 20dB attenuation.

17. **Output :** 50Ω BNC socket of the Tracking Generator. The output level can be determined from + dB to -50 dBm.

18. **TRCK. GEN. :** The Tracking generator is activated if the push button is depressed (ON). In this case a sine signal can be obtained from the OUTPUT BNC socket with a frequency determined by the Spectrum Analyzer. In ZERO SCAN mode the center Frequency appears at the output.

19. **Level :** The output level of the Tracking can be continuously adjusted with this knob by 11dBm (-10 dB to +1dBm).

20. **X-POS:** X-Position adjustment , (screwdriver)

21. **X-Ampl:** X gain adjustment , (screwdriver adjustment , to be carried out by qualified engineer with reference source availability)

22. **Phone :** An earphone or loudspeaker with an impedance 16Ω can be connected to this output. When tuning the Spectrum Analyzer to a spectral line possibly availably available audio signals can be detected. The signal line is provided by an AM-Demodulator in the IF – section. It demodulates any available AM-Signal an provides as well one-side FM-Demodulation. The output is short circuit proof.

23. **Volume :** Volume setting for earphone output.

24. **Probe Power:** The output provides a +6 Vdc voltage for the operation of an optional near field sniffer probe. It is only provided for this purpose and requires a special cable which is shipped along with the probe set.

Operation

General information

The logical front panel layout of INSA10 ensures rapid familiarization with the various function. However even experienced operator should not neglect to carefully read the following instructions. To avoid any operational errors and to be fully acquainted with the instrument when later in use.

Use of tilt handle

To view the screen from the best angle there are three different positions for setting up the instrument. If the instrument is set down on the floor after being carried the handle remains automatically in the upright carrying position.

Operating Conditions

The ambient temperature range during operation should be +0 to 40 o C RH 95% should not exceed -40o C or +70 o C during storage

Introduction to Spectrum Analyzer

The Spectrum analyzer permits the detection of spectrum components of electrical signal in the frequency range of 0.15 to 1050MHz. The detected signal and its content have to be repetitive. in contrast to an Oscilloscope operated in Yt mode. Where the amplitude is displayed on the time domain the spectrum analyzer displays amplitude on the frequency domain Yf. The individual spectrum component of signal become visible on a spectrum analyzer. The Oscilloscope display the same signal has one resulting wave form.

The spectrum analyzer works according to the supperhetrodyne receiver principle. The signal to be measured (fin = 0.15 to 1050MHz) is applied to the first mixer it is mixed with the signal of a variable voltage controlled oscillator(LO 1350 to 2350MHz). The oscillator is called the first LO. The different between the oscillator and the input frequency is the first intermediate frequency which passes through a wave band filter tuned to a center frequency of 1350MHz. It then enters an amplifier and this is followed by two additional mixing stages oscillators and amplifiers. The second IF is 29.875MHz and the third is 2.75MHz. in the third IF stage, the signal can be selectively transferred through a filter with 400kHz or 20 kHz bandwidth before arriving at an AM demodulator. The logarithmic output (video signal) is transferred directly osr via a low pass filter to another amplifier. This amplifier output is connected to the Y deflection plates of the CRT.

The X deflection is performed with a ramp generator voltage. This voltage can also be superimposed on a dc voltage which allows for the control of 1st LO. The spectrum analyzer

scans a frequency range depending on the ramp height. This span is determined by the scanwidth setting in ZERO SCAN mode only direct voltage controls the 1st LO.
The INSA10 also includes a tracking generator. This generator provides sine wave voltage within the frequency range of 0.15 to 1050MHz. the tracking generator is determined by the first oscillator (1st LO) of the spectrum analyzer section. Spectrum analyzer and tracking generator are frequency synchronized.

Operating Instructions

The most sensitive component of the INSA10 is the input section of the Spectrum Analyzer. It consists of the signal attenuator and the first mixer. Without input attenuation , the voltage at the input must not exceed +10 dB (0.7Vrms) AC or ±25 volt DC. With a maximum input attenuation of 40 dB the AC voltage the must not exceed +20 dBm. These limits must not be exceeded. Otherwise the input attenuator and/or the first mixer would be destroyed. When measuring via a LISN (line impedance stabilization network) the input of the Spectrum Analyzer must be protected by means of a transient limiter.
The user should also consider the possibility of excessively high signal amplitudes outside the covered frequency range although not displayed (e.g. 1200MHz). The frequency range of 0 Hz 150kHz is not specified for the INSA10 Spectrum Analyzer. Spectral lines within this range would be displayed with incorrect amplitude. A particularly high intensity setting shall be avoided. The way signals are displayed on the spectrum analyzer typically allows for any signal to be recognized easily, even with low intensity. Due to the frequency conversion principle, a spectral line is visible at 0 Hz. It is called If-feed through. The line appears when the 1st LO frequency passes the If amplifiers and filters. The level of this spectral line is different in each instrument. A deviation from the full screen does not indicate a malfunctioning instrument.

Vertical Calibration

Ensure all input attenuators in released position before calibration. When the signal amplitude reaches to the highest level-line, the signal amplitude is -27 dBm, and it will reduce 10 dBm for each division drop. If the 40 dB attenuator in the spectrum analyzer is completely pressed down, the highest level amplitude will be +13 dBm (-27 dBm +40 dBm).
Prior to calibration ensure that all input attenuators (14) are released. The INSA10 must be in operation for at least 60 minutes prior to calibration. Switch VIDEO FILTER (11) to OFF position set BANDWIDTH (10) to 400kHz, and SCANWIDTH (15) to the spectrum analyzer input (13). The frequency of this signal should be 500MHz set the center frequency to the signal frequency.

Horizontal Calibration

Prior to calibration ensure that all input attenuator switches (14) are released. The INSA10 must be operated for at least 60 minutes prior to calibration. The VIDEO FILTER push button (11) must be in OFF position the BANDWIDTH (10) must be set to 400 kHz, and SCANWIDTH (15) set to 100kHz/div. After the center frequency is set to 500 MHz. a generator signal must be applied to the input. The output level should be between 40 and 50 dB above the noise.Set generator frequency to 500 MHz. Adjust the peak of the 500 MHz spectral line to the horizontal screen center using the X-POS controlSet the generator frequency to 100 MHz. If the 100MHz Control.

Introduction to Spectrum Analysis

The analysis of electrical signals is a fundamental problem for many engineers and scientists. Even if the immediate problem is not electrical, the basic parameters of interest are often changed into electrical signals by means of transducers. The rewards for transforming physical parameters to electrical signals are great, as many instruments are available for the analysis of electrical signals in the time and frequency domains.The traditional way of observing electrical signals is to view them in the time domain using an oscilloscope. The time domain is used to recover relative timing and phase information which is needed to characterize electric circuit behavior. However , not all circuits can be uniquely characterized from just time domain information. Circuit elements such as amplifiers, oscillators, mixers, modulators, detectors and filters are best characterized by their frequency response information. This frequency information is best obtained by viewing electrical signals in the frequency domain. To display the frequency domain requires a device that can discriminate between frequencies while measuring the power level at each. One instrument which displays the frequency domain is the spectrum analyzer. It graphically displays voltage or power as a function of frequency only on a CR T. In the time domain frequency components of a signal are seen summed together in the frequency domain complex spectral line is not on the 2nd graticule line from left, it should be aligned using the X-AMP signals (i.e. signals composed of more than one frequency) are separated into their frequency components, and the power level at each frequency is displayed. The frequency domain is a graphical representation signal amplitude as a function of frequency. The frequency domain contains information not found in the time domain and therefore the spectrum analyzer has certain advantages compared with an oscilloscope.

The analyzer is more sensitive to low level distortion than a scope. Sine waves may look in the time domain but in the frequency domain harmonic distortion can be seen. The sensitivity and wide dynamic range of the spectrum analyzer is useful for measuring low level modulation. It can be used to measure AM, FM and pulsed RF The analyzer can be used to measure carrier frequency , modulation frequency, modulation distortion.

The spectrum analyzer can be used to measure long and short term stability. Parameters such as noise sidebands on an oscillator, residual FM of a source and frequency drift during warm-up can be measured using the spectrum analyzer‟s calibrated scans. The swept frequency responses of a filter or amplifier are examples of swept frequency measurements possible with a spectrum analyzer. These measurements are simplified by using a tracking generator.

Types of Spectrum Analyzer

There are two basic types of spectrum analyzers swept-tuned and real-time analyzer. The swept-tuned analyzers are tuned by electrically sweeping them over their frequency range. Therefore, the frequency components of a spectrum are sampled sequentially in time. This enables periodic and random signals to be displayed.

Spectrum Analyzer Requirements

button (11) must be in OFF position the BANDWIDTH (10) must be set to 400 kHz, and SCANWIDTH (15) set to 100kHz/div. After the center frequency is set to 500 MHz. a generator signal must be applied to the input. The output level should be between 40 and 50 dB above the noise.Set generator frequency to 500 MHz. Adjust the peak of the 500 MHz spectral line to the horizontal screen center using the X-POS controlSet the generator frequency to 100 MHz. If the 100MHz spectral line is not on the 2nd graticule line from left, it should be aligned using the X-AMP Control.

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Spectrum Analyzer Requirements

To accurately display the frequency and amplitude of a signal on a spectrum analyzer, the analyzer itself must be properly calibrated. A spectrum analyzer properly designed for accurate frequency and amplitude measurements has to satisfy many requirements:

- Wide tuning range
- Wide frequency display range
- Stability
- Resolution
- Flat frequency
- High sensitivity
- Low internal distortion

Frequency Measurement

The frequency scale can be scanned in three different modes full, per division, and zero scan The full scan mode is used to locate signals because the widest frequency ranges are displayed in this mode. (not all spectrum analyzers offer this mode). The per division mode is used to zoom-in on a particular signal. In per division the center frequency of the display is

set by the tuning control and the scale factor is set by the frequency span of scan width control. In the zero scan mode, the analyzer acts as a fixed-tuned receiver with selectable bandwidths.
Absolute frequency measurements are usually made from the spectrum analyzer tuning dial. Relative frequency measurements require a linear frequency scan. By measuring the relative separation of two signals on the display the display the frequency difference can be determined.
It is important that the spectrum analyzer be more stable than the signals being measured. The stability of the analyzer depends on the frequency stability of its local oscillators. Stability is usually characterized as either short term or long term. Residual FM is a measure of the short term stability which is usually specified in Hz peak-to-peak. Short term stability is also characterized by noise sidebands which are a measure of the analyzers spectral purity. Noise sidebands are specified in terms of dB down and Hz away from a carrier in a specific bandwidth. Long term stability is characterized by the frequency drift of the analyzers loss. Frequency drift is measure of how much the frequency changes during a specified time(i.e.Hz/hr)

Resolution

Before the frequency of a signal can be measured on a spectrum analyzer it must first be re-solved. Resolving a signal means distinguishing it from its nearest neighbors. The resolution of a spectrum analyzer is determined by its IF bandwidth. The IF bandwidth is usually the 3 dB bandwidth of the IF filter. The ratio of the 60 dB bandwidth (in Hz) to the 3 dB bandwidth(in Hz) is known as the shape factor of the filter. The smaller the shape factor , the greater is the analyzer capability to resolve closely spaced signals of unequal amplitude. If the shape factor of a filter is 15: I then two signals whose amplitudes differ by 60 dB must differ in frequency by 7.5 time the IF bandwidth before they can be distinguished separately. Otherwise, they will appear as one signal on the spectrum analyzer display. The ability of a spectrum analyzer to resolve closely spaced signals of unequal amplitude is not a function of the IF filter shape factor only. Noise sidebands can also reduce the resolution. They appear above the skirt of the IF filter and reduce the off band rejection of the filter. This limits the resolution when measuring.
The resolution of the spectrum analyzer is limited by its narrowest IF bandwidth. For example, if the narrowest bandwidth is 10kHz then the nearest any two signals can be and still be resolved is 10kHz. This is because the analyzer traces out its own IF band-pass shape as it sweeps through a CW signal. Since the resolution of the analyzer is limited by bandwidth it seems that by reducing the IF bandwidth infinitely, infinite resolution will be achieved. The fallacy here is that the usable IF bandwidth is limited by the stability (residual Fm) of the analyzer. If the internal frequency deviation of the analyzer is 10 kHz, then the narrowest bandwidth that can be used to distinguish a single input signal is 10kHz. Any narrower IF-filter will result in more than one response or an intermittent response for a single input frequency. A practical limitation exists on the IF bandwidth as well, since narrow filters have ling time constants and would require excessive scan time.

Sensitivity

Sensitivity is a measure of the analyzer‟s ability to detect small signals. The maximum sensitivity of an analyzer is limited by its internally generated noise. The noise is basically of two types: thermal (or johnson) and non thermal noise. Thermal noise power can be expressed as:
Pn= K. T. B
Where

Pn = Noise power in watts
K =Boltzmanns constant($1.38\times10-23$ Joule/K) T = absolute temperature, K
B = bandwidth of system is Hertz
As seen from this equation the noise level is directly proportional to bandwidth. Therefore a decade decrease in bandwidth results in a 10 dB decrease in noise level and consequently 10 dB better sensitivity. Non thermal noise accounts for all noise produced within the analyzer that is not temperature dependent. Spurious emissions due to nonlinearities of active elements impedance mismatch, etc. are sources of non thermal noise. A figure of merit or noise figure, is usually assigned to this non thermal noise which when added to the thermal noise gives the total noise of the analyzer system. This system noise which is measured on the CRT, determines the maximum sensitivity of the spectrum analyzer. Because noise level changes with bandwidth it is important, when comparing the sensitivity of two analyzers. To compare sensitivity specifications for equal bandwidths. A spectrum analyzer sweeps over a wide frequency range but is really a narrow band instrument. All of the signals that appear in the frequency range of the analyzer are converted to a single IF frequency which must pass through an IF filter, the detector sees only this noise at any time. Therefore the noise displayed on the analyzer is only that which is contained in the IF passband. When measuring discrete signals, maximum sensitivity is obtained by using the narrowest IF bandwidth.

Video Filter

Measuring small signals can be difficult when they are approximately the same amplitude as the average internal noise level of the analyzer. To facilitate the measurement. It is best to use video filtering. A video filter is a post- detection low pass filter which averages the internal noise of the analyzer. When the noise is averaged the input signal may be seen. If the resolution bandwidth is very narrow for the span the video filter should no be selected as this will not allow the amplitude of the analyzed signals to reach full amplitude due to its video bandwidth limiting property.

Spectrum Analyzer Sensitivity

Specifying sensitivity on a spectrum analyzer is somewhat arbitrary. One way of specifying sensitivity is to define it as the signal level when signal power = average noise power. The analyzer always measures signal plus noise. Therefore when the input signal is equal to the internal noise level, the signal will appear 3dB above the noise. When the signal power is added to the average noise power the power level on the CRT is doubled (increased by 3dB) because the signal power = average noise power.

The maximum input level of the spectrum analyzer is the damage level or bum- out level of the input circuit. This is +10 dB for the input mixer and +20dB for the input attenuator.

Before reaching the damage level of the analyzer the analyzer will begin to gain compress the input signal. This gain compression is not considered serious until it reaches 1 dB. The maximum input signal level which will always result in less than 1 dB gain compression is called the linear input level. Above 1 dB gain compression the analyzer is considered to be operating nonlinearly because the signal amplitude displayed in the CRT is not an accurate measure of the input signal level.

Whenever a signal is applied to the input of the analyzer, distortions are produced within the analyzer itself. Most of these are caused by the non-linear behavior of the input mixer. For the INSA 10 these distortions are typically 70 dB below the input signal level for signal levels not exceeding -27dBm at the input of the first mixer. To accommodate larger input signal levels, an attenuator is placed in the input circuit before the first mixer. The largest input signal that can be applied at each setting of the input attenuator, while maintaining the internally generated distortions below a certain level, is called the optimum input level of the analyzer. The signal is attenuated before the first mixer because the input to the mixer must not exceed -27dB, or the analyzer distortion products may exceed the specified 70dB range. This 70Db distortion-free range is called the spurious- free dynamic range of the analyzer. The display dynamic range is defined as the ratio of the largest signal to the smallest signal that can be displayed simultaneously with no analyzer distortions present.

Dynamic range requires several things then. The display range must be adequate no spurious or unidentified response can occur and the sensitivity must be sufficient to eliminate noise from the displayed amplitude range.

The maximum dynamic range for a spectrum analyzer can be easily determined from its specifications. First check the distortion spec. for example this might be “all spurious products 70dB down for -27 dBm at the input mixer”. Then determine that adequate sensitivity exists. For example. 70dB down from -27dBm is -97dB. This is the level we must be able to detect and the bandwidth required for this sensitivity must not be too narrow or it will be useless. Last the display range must be adequate.

Notice that the spurious –free measurement range can be extended by reducing the level at the input mixer. The only limitation then is sensitivity. To ensure a maximum dynamic range on the CRT display, check to see that the following requirements are satisfied.

• The largest input signal does not exceed the optimum input level of the analyzer (typically -27 dBm with 0dB) input attenuation.

• The peak of the largest input signal rests at the top of the CRT display(reference level).

Frequency Response

The frequency response of an analyzer is the amplitude linearity of the analyzer over its frequency range. If a spectrum analyzer is to display equal amplitudes for input signals of equal amplitude, independent of frequency then the conversion (power) loss of the input mixer must not depend on frequency. If the voltage from the LO is too large compared to the input signal voltage then the conversion loss of the input mixer is frequency dependent and the frequency response of the system is nonlinear. For accurate amplitude measurements, a spectrum analyzer should be as flat as possible over its frequency range. Flrtness is usually the liming factor in amplitude accuracy since its extremely difficult to calibrate out. And since the primary function of the spectrum analyzer is to compare signal level at different frequencies, a lack of flatness can seriously limit its usefulness,

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EXPERIMENT NO-6
PRE-EMPHASIS & DE-EMPHASIS

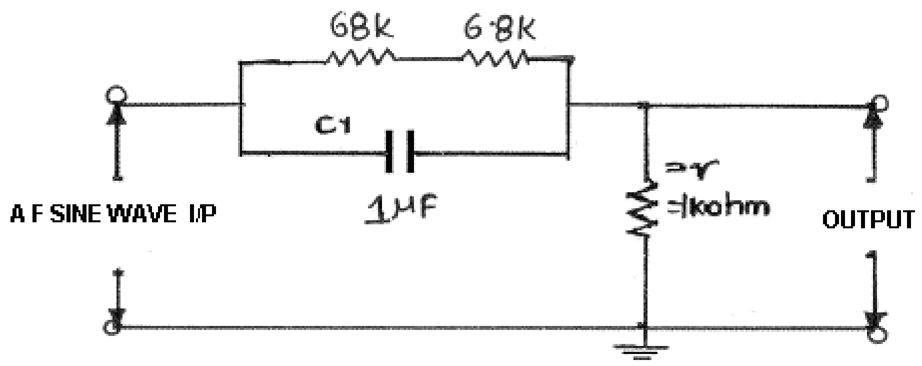
AIM: To study the functioning of Pre-Emphasis and De-Emphasis circuits.

APPARATUS:

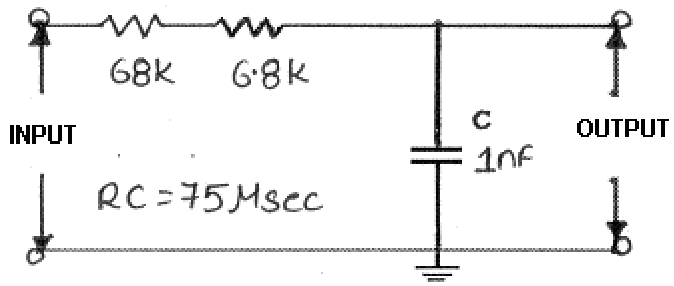
1.Pre-emphasis & De-emphasis trainer kits.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Patch chords and Probes.

CIRCUIT DIAGRAM:

PRE EMPHASIS CKT



DE EMPHASIS CIRCUIT



THEORY:

Frequency modulation is much more immune to noise than amplitude modulation and is significantly more immune than phase modulation. The threshold effect is more serious in FM as compared to AM, because in FM, the signal to noise ratio at the input of a detector, at which threshold effect starts, is higher. Lower the threshold level, better is the system because threshold can be avoided at a comparatively lower ratio, and a small signal is needed to avoid threshold for an equivalent noise power. Hence, it is desirable to lower the threshold level in the FM receivers. The process of lowering the threshold level is known as threshold

improvement, or threshold reduction. Two methods are used for the improvement of the threshold.
1. Pre-Emphasis and De-Emphasis circuits.
2. FMFB (Frequency Modulation with Feed Back.)

PRE-EMPHASIS AND DE-EMPHASIS:
The noise triangle shows, noise has a greater effect on the higher modulating frequencies than on the lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, there by increasing the signal-to-noise ratio. This boosting of the higher modulating frequencies, in accordance with a prearranged curve, is termed pre-emphasis, and the compensation at the receiver is called de-emphasis.

PROCEDURE:

PRE-EMPHASIS:

1. Connect the circuit as per the circuit diagram
2. Apply a sine wave to the input terminals of 2 VP-P (Vi)
3. By varying the input frequency in steps from 30HZ to 15KHZ with fixed amplitude, note down the output amplitude (Vo) with respect to the input frequency.
4. Calculate the gain using the formula

Gain = 20 log (VO/ VI) db
Where VO = output voltage in volts.
VI = Input voltage in volts.
Normalized gain (in dB) at any frequency - gain (in dB) at 500HZ.
5. Plot the frequency response.

DE-EMPHASIS:
Repeat the above procedure for De emphasis ckt.
OVER ALL RESPONSE:

1. Connect the output of the pre-emphasis circuit to the input of de-emphasis circuit.
2. Give the input from signal generator to the pre emphasis circuit.
3. Repeat steps 3,4 & 5 of Pre-Emphasis to de-emphasis also.

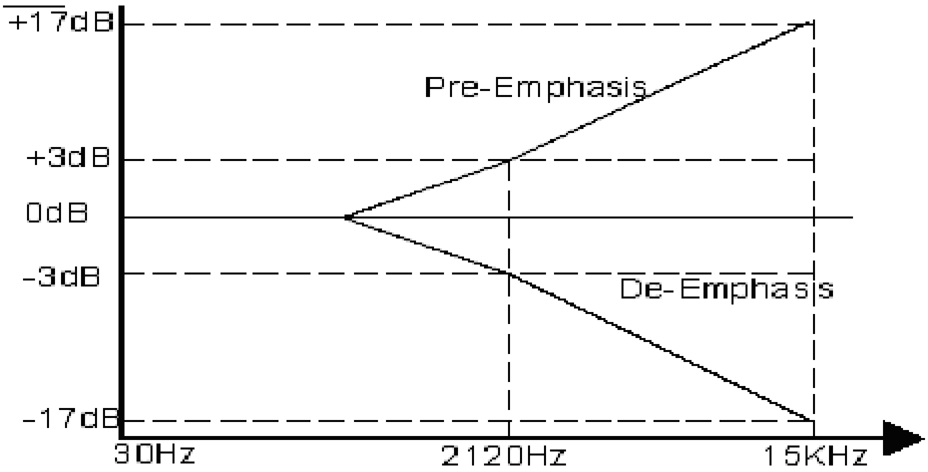
TABULAR COLUMN FOR PRE EMPHASIS

S.No	Input Frequency (30Hz to 15KHz)	Out put voltage (Vo) (volts)	GAIN 20 log (V_O/ V_I) db	Normalized gain(dB)

TABULAR COLUMN FOR DE EMPHASIS

S.No	Input Frequency (30Hz to 15KHz)	Out put voltage (Vo) (volts)	GAIN 20 log (V_O/ V_I) db	Normalized gain(dB)

EXPECTED WAVEFORMS:



RESULT:.

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QUESTIONS

1.What is the need for pre-emphasis?
2.Explain the operation of pre-emphasis circuit?
3.Preemphasis operation is similar to high pass filter explain how?
4.Deemphasis operation is similar to low pass filter justify?
5.What is de-emphasis?
6.Draw the frequency response of a pre-emphasis circuit?
7. Draw the frequency response of a de-emphasis circuit?
8. Give the formula for the cutoff frequency of the pre-emphasis circuit?
9.What is the significance of the 3db down frequency?

MATLABCODE

```
fs=5000;   Ts=1/fs;   fm1=20;
fm2=30;  fc=200;
t=(0:num_samples-1)*Ts;
f=(-num_samples/2:num_samples/2-1)*fs/num_samples;
mt=sin(2*pi*fm1*t);
Mf=fftshift(abs(fft(mt));
f_cutoff_pe=15;
Wn_pe=f_cutoff_pe/(fs/2);
[b_pe,a_pe]=butter(1,Wn_pe);
[H_pe,W]=freqz(a_pe,b_pe);
a_de=b_pe;

b_de=a_pe;
[H_de,W]=freqz(a_de,b_de);
mt_pe=filter(a_pe,b_pe,mt);
Mf_pe=fftshift(abs(fft(mt_pe)));
figure(1)
subplot(211);plot(t,mt)
axis([0.6min(mt)-1 max(mt)+1])
grid on;title('Modulating Signal (Time Domain)')
subplot(212);plot(f,Mf)
grid on;axis([-50 50 0 max(Mf)+100])
title('Modulating Signal (Frequency Domain)')
figure(2)
subplot(211)
semilogx(W*pi*(fs/2),abs(H_pe),'m','linewidth',2)
axis([0 fs/2 0 50])
grid on;title('Pre-emphasis Filter Magnitude Response')
subplot(212)
semilogx(W*pi*(fs/2),abs(H_pe),'m','linewidth',2)
axis([0 fs/2 0 1])
grid on;title('De-emphasis Filter Magnitude Response')
figure(3)
subplot(211);plot(t,mt_pe,'r')
axis([0.6min(mt_pe)-1 max(mt_pe)+1])
grid on;title('Pre-emphasized Signal (Time Domain)')
subplot(212);plot(f,Mf_pe,'r')
grid on;axis([-50 50 0 max(Mf_pe)/5+100])
title('Pre-emphasized Signal (Frequency Domain)')
freqdev=8;
Max_Dev=freqdev*max(mt);
Min_Dev=freqdev*min(mt);
Max_Dev_pe=freqdev*max(mt_pe);
Min_Dev_pe=freqdev*min(mt_pe);
len=size(mt,1);
if(len==1)
mt=mt(:);
```

```
end
t=(0:1/fs:((size(mt,1)-1)/fs));
t=t(:ones(1,size(mt,2)));
int_mt_pe=cumsum(mt_pe)/fs;
mt_pe_FM=cos(2*pi*fc*t+2*pi*freqdev*int_mt_pe);
mt_FM_spec=fftshift(abs(fft(mt_FM)));
mt_pe_FM_spec=fftshift(abs(fft(mt_pe_FM)));
figure(4)
subplot(221)plot(t,mt_FM)
axis([0.6min(mt_FM)-1 max(mt_FM)+1])
grid on;title('FM output without Pre-emphasis (Time Domain)')
subplot(222);plot(f,mf_FM_spec)
grid on;axis([-300 300 0 max(mf_FM_spec)+50])
title(' FM output without Pre-emphasis (Frequency Domain)')
subplot(223);plot(t,mt_pe_FM,'r')
axis([0.6min(mt_pe_FM)-1 max(mt_pe_FM)+1])
grid on;title('FM output with Pre-emphasis (Time Domain)')
subplot(224);plot(f,mt_pe_FM_spec,'r')
grid on;axis([-300 300 0 max(mt_pe_FM_spec)+50])
title(' FM output with Pre-emphasis (Frequency Domain)')

xt1=awgn(mt_FM,10,'measured');
xf1=fftshift(abs(fft(xt1)));
wt=xt1-mt_FM;
xt2=mt_pe_FM+wt;
xf2=fftshift(abs(fft(xt2)));
figure(5)
subplot(221);plot(t,xt1)
axis([0.6min(xt1)-1 max(xt1)+1])
grid on;title('Noisy FM without Pre-emphasis (Time Domain)')
subplot(222);plot(f,xf1)
grid on;axis([-300 300 0 max(xf1)+50])
title(' Noisy FM without Pre-emphasis (Frequency Domain)')
subplot(223);plot(t,xt2,'r')
axis([0.6min(xt2)-1 max(xt2)+1])
grid on;title('Noisy FM with Pre-emphasis (Time Domain)')
subplot(224);plot(f,xf2,'r')
grid on;axis([-300 300 0 max(xf2)+50])
title(' Noisy FM with Pre-emphasis (Frequency Domain)')
N=5;
f_cutlow=fc-50;
f_cuthigh=fc+50;
Wn1=f_cutlow/(fs/2);
Wn2=f_cuthigh/(fs/2);
Wn_BPE=[Wn1Wn2];
[a_BPE,b_BPE]=butter(N,Wn_BPE);
yt1=filtfilt(a_BPE,b_BPE,xt1);
yt1=fftshift(abs(fft(yt1)));
```

```
f_cutlow_2=fc+min_Dev_pe;
f_cuthigh_2=fc+max_Dev_pe;
Wn1_2=f_cutlow_2/(fs/2);
Wn2_2=f_cuthigh_2/(fs/2);
Wn_BPE2=[Wn1_2 Wn2_2];
[a_BPE2,b_BPE2]=butter(N,Wn_BPE);
yt2=filtfilt(a_BPE2,b_BPE2,xt2);
yt1=fftshift(abs(fft(yt2)));
f_cut=fm2+5;
Wn_LPF=f_cut/(fs/2);
[a_LPF,b_LPF]=butter(N,Wn_LPF);
t=(0:1/fs:((size(yt1,1)-1)/fs));
t=t(:ones(1,size(yt1,2)));
yq=hilbert(yt1).*exp(-j*2*pi*fc*t);
rt1=(1/(2*pi*freqdev))*[zeros(1,size(yq,2));diff(unwrap(angle(yq)))*fs];
rt1=filter(a_LPF,b_LPF,rt1);
Rf1=fftshift(abs(fft(rt1)));
t=(0:1/fs:((size(yt2,1)-1)/fs));
t=t(:ones(1,size(yt2,2)));
yq=hilbert(yt2).*exp(-j*2*pi*fc*t);
rt2=(1/(2*pi*freqdev))*[zeros(1,size(yq,2));diff(unwrap(angle(yq)))*fs];
rt2=filter(a_LPF,b_LPF,rt2);
Rf2=fftshift(abs(fft(rt2)));
rt_de=filter(a_de,b_de,rt2);
Rf_de=fftshift(abs(fft(rt_de)));
figure(6)
subplot(221);plot(t,yt1)
axis([0.6min(yt1)-1 max(yt1)+1])

grid on;title('FM BP Filtered data (Time Domain)')
subplot(222);plot(f,yf1)
grid on;axis([-300 300 0 max(yf1)+50])
title('FM BP Filtered data (Frequency Domain)')
subplot(223);plot(t,yt2,'r')
axis([0.6min(yt2)-1 max(yt2)+1])
grid on;title('FM BP Filtered data with PE (Time Domain)')
subplot(224);plot(f,yf2,'r')
grid on;axis([-300 300 0 max(yf2)+50])
title('FM BP Filtered data with PE (Frequency Domain)')
figure(7)
subplot(321);plot(t,rt1)
axis([0.6min(rt1)max(rt1)])
grid on;title('FM Receiver output without PE-DE(Time Domain)')
subplot(322);plot(f,Rf1)
grid on;axis([-50 50 0 max(Rf1)+50])
title('FM Receiver output without PE-DE(Frequency Domain)')
subplot(323);plot(t,rt2,'r')
axis([0.6min(rt2)max(rt2)])
```

```
grid on;title('FM Receiver output with PE(Time Domain)')
subplot(324)
plot(f,Rf2)
grid on;axis([-50 50 0 max(Rf2)+50])
title('FM Receiver output with PE (Frequency Domain')
subplot(325);plot(t,rt_de,'r')
axis([0.6min(rt_de)max(rt_de)])
grid on
title('FM Receiver output with PE-DE(Time Domain)')
subplot(326);plot(f,Rf_de)
grid on;axis([-50 50 0 max(Rf_de)+50])
title('FM Receiver output with PE-DE(Frequency Domain')
```

SIMULATED OUTPUT WAVEFORMS

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ATTACH ONE GRAPH SHEET HERE

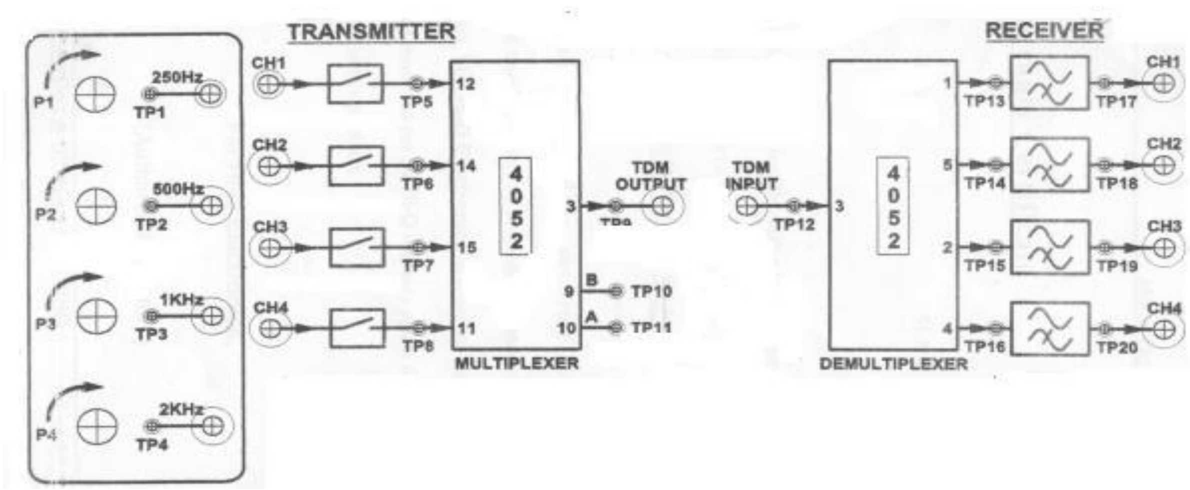
EXPERIMENT NO-7

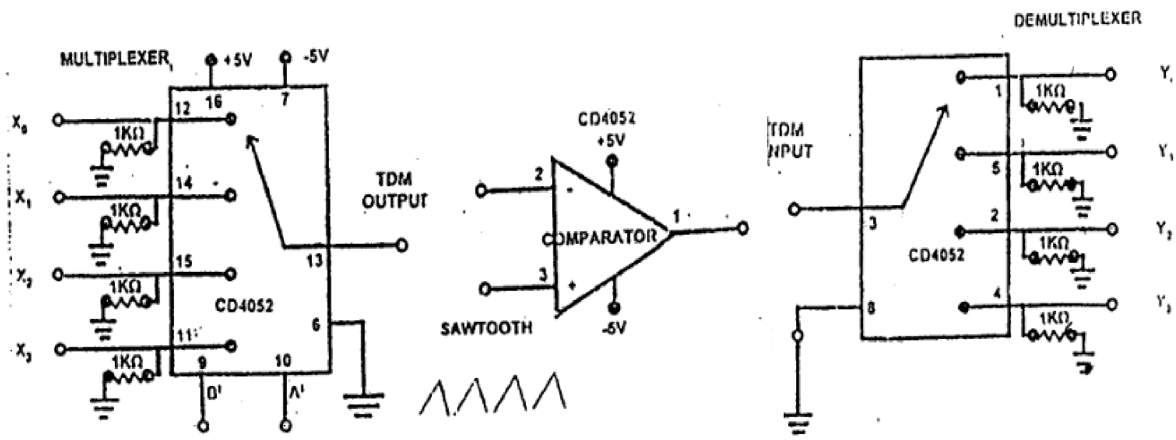
TIME DIVISION MULTIPLEXING & DEMULTIPLEXING

AIM:
1. Study of 4 Channel Analog Multiplexing and Demultiplexing Techniques.
2. Study of the effect of sampling frequency variation on the output.
3. Study of input signal amplitude on the output.

APPARATUS:
1.TIME DIVISION MULTIPLEXING & DEMULTIPLEXING Trainer.
2.C.R.O (30 MHz)
3.Patch chords.

CIRCUIT DIAGRAM:





THEORY:-

The TDM is used for Transmitting several analog message signals over a communication channel by dividing the time frame into slots, one slot for each message signal. The four input signals, all band limited by the input filters are sequentially sampled, the output of which is a PAM waveform containing samples of the input signals periodically interlaced in time. The samples from adjacent input message channels are separated by Ts/M, where M is the number of input channels. A set of M pulses consisting of one sample from each of the input M-input channels is called a frame.

At the receiver the samples from individual channels are separated by carefully synchronizing and is critical part TDM. The samples from each channel are filtered to reproduce the original message signal. There are two levels of synchronization. Frame synchronization is necessary to establish when each group of samples begin and word synchronization is necessary to property separate the samples within each frame. Besides the space diversity & frequency diversity there is a method of sending multiple analog signals on a channel using "TIME DIVISION MULTIPLEXING & DEMULTIPLEXING" Technique

Circuit Description:-

Function Generator Circuit:-

A 4.096 MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comprising a 4.096MHz crystal and three 74HC04(U2) inverter gates. this 4.096 MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U3), to produce 50% duty cycle, 64 KHz square wave on pin no.1 of U4, and 32KHz square wave on pin no.4. 32 KHz square wave is given to pin no.2 of IC NE555(U7) which act as a monostable multivibrator. Potentiometer P5 adjust the pulse width. 64KHz square wave is fed to the four bit binary counter on pin no.1 to produce 4KHz square wave at pin no.6. this goes to pin no.13. this signal clocks to the second half of the counter to produce square wave at following frequencies.

Each of these square wave outputs is then fed to its own low pass filter circuits TL072(U8,U9). Which generates corresponding sine wave outputs. The amplitude of this sine wave can be varied by potentiometers P1,P2,P3,P4 respectively. These sine wave outputs are available at TP1,TP2,TP3,TP4 respectively and have amplitudes upto 10V max.

Transmitter Block:-

Each modulating signal is applied to IC TLO74(U6)(pin nos. 3,5,10,12 respectively). This IC buffers the applied signal and is fed to pin nos.3,14,11,6 respectively of IC DG211(U5). The pulse input(32KHz(clock)) is applied to pin nos. 1,16,9,8 of U5. Corresponding PAM outputs are available at test points Tp5,Tp6,Tp7,Tp8 respectively. These each PAM outputs are applied to IC 4052(U1). Which is act as multiplexer.

4 Channel Demultiplexer:-

The multiplexed PAM signal is given to the 4 channel Demultiplexer input at pin 3(TP12). The A& B timing wave forms selects the channel and accordingly connects the same to the output. This at the PAM signal of each channel are separated these separated demultiplexed outputs are monitored at test points 13,14,15,16 respectively.

Low Pass Filter:-

Each separated PAM outputs are being connected to corresponding channel‟s butter worth Low Pass Filter. This is 4th order filter having roll of rate of 24db/octave(40db/decade) and cut-off frequency of 250Hz,500Hz,1KHz,2KHz respectively. The output of these filters goes to corresponding sockets termed as CH1,CH2,CH3,CH4. The reconstructed signal can be monitored at test points 17,18,19,20 respectively. These outputs are at lower amplitude.

PROCEDURE

1.Connect the adaptor to the mains.

2. Adjust the frequency of 555 timer to be around 32KHz, so that each of the 4channels is sampled at 8KHz.

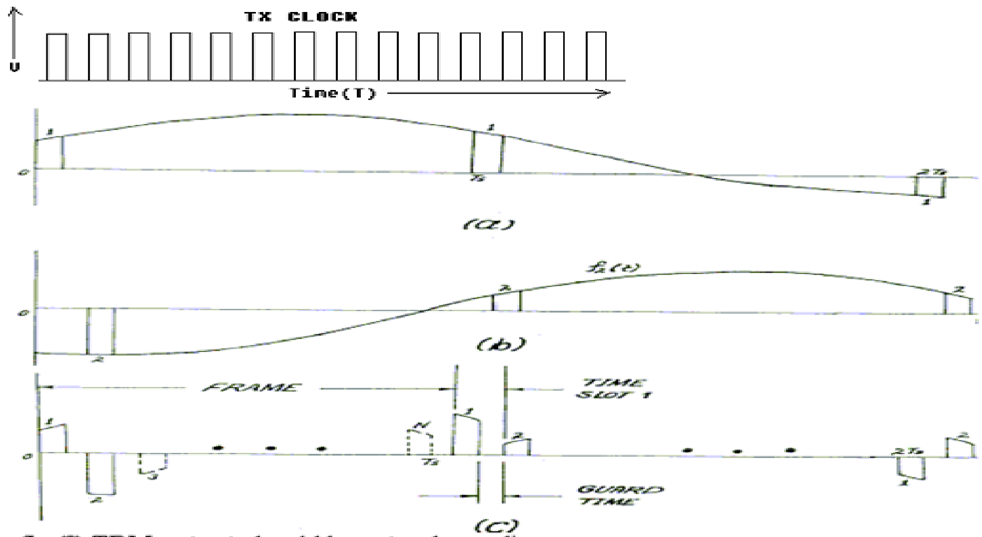
3. Observe output at pin 11 of 7476 on CH1 and TDM output at pin 13 of CD4052 on CH2 of CRO. All multiplexed channels are observed during the full period of the clock.

4. Connect the TDM output pulses to TDM input of de multiplexer at pin 13 of CD4052. Observe individual outputs 1,2,3,4 at pin 12,14,15,11 of CD4052 respectively. The TDM pulses corresponding to each channel are now separated as 4 streams.

5. Take any output of the demodulator and connect it to filter and observe the output of the amplifier in conjuction with the corresponding input. Repeat this for all inputs.

6.Vary the frequency of sampling by potentiometer P1 connected to 555 and observe the effect on the individual effects.

EXPECTED WAVEFORMS



fig(2) TDM output should be natural sampling
Fig(a) message1 (b) message 2 and

RESULT :

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QUESTIONS

1.Draw the TDM signal with 2 signals being multiplexed over the channel?
2.Define guard time & frame time?
3.Explain block schematic of TDM?
4.How TDM differ from FDM?
5.What type of filter is used at receiver end in TDM system?
6.what are the applications of TDM?
7.If 2 signal band limited to 3 kHz, 5KHz & are to be time division multiplexed. What is the maximum permissible interval between 2 successive samples.?
8.Is the bandwidth requirement for TDM & EDM will be same?
9.Is the circuitry needed in FDM.?
10.Is TDM system is relatively immune to interference with in channels (inter channel cross

MATLAB CODE:

```
clc;
close all;
clear all;
% Signal generation
x=0:.5:4*pi; % siganal taken upto 4pi
sig1=8*sin(x); % generate 1st sinusoidal signal
l=length(sig1);
sig2=8*triang(l); % Generate 2nd traingular Sigal
% Display of Both Signal
subplot(2,2,1);
plot(sig1);
title('Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,2,2);
plot(sig2);
title('Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Display of Both Sampled Signal
subplot(2,2,3);
stem(sig1);
title('Sampled Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,2,4);

stem(sig2);
title('Sampled Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
l1=length(sig1);
l2=length(sig2);
for i=1:l1
sig(1,i)=sig1(i); % Making Both row vector to a matrix
sig(2,i)=sig2(i);
end
% TDM of both quantize signal
tdmsig=reshape(sig,1,2*l1);
% Display of TDM Signal
figure
stem(tdmsig);
title('TDM Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Demultiplexing of TDM Signal
demux=reshape(tdmsig,2,l1);
```

```
for i=1:l1
sig3(i)=demux(1,i); % Converting The matrix into row vectors
sig4(i)=demux(2,i);
end
% display of demultiplexed signal
figure
subplot(2,1,1)
plot(sig3);
title('Recovered Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,1,2)
plot(sig4);
title('Recovered Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
```

SIMULATED OUTPUT WAVEFORMS

SIGNATURE OF THE LAB INCHARGE

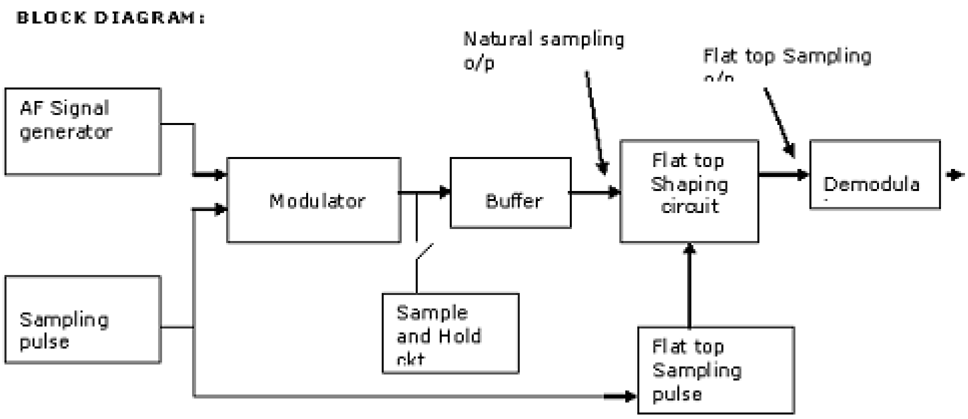
ATTACH ONE GRAPH SHEET HERE

EXPERIMENT NO-8
SAMPLING AND RECONSTRUCTION

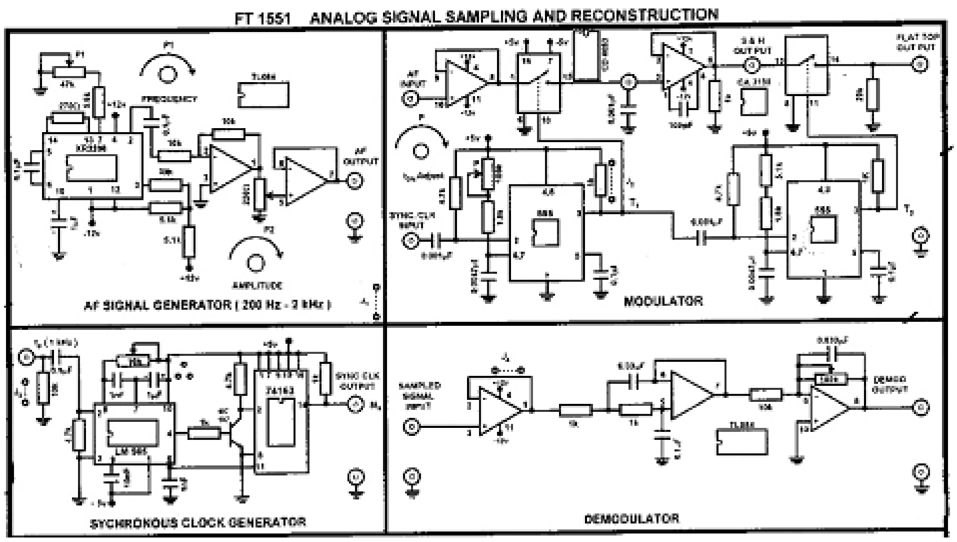
AIM:
1. To study the sampling theorem and its reconstruction.
2. To study the effect of amplitude and frequency variation of modulating signal on the output.
3. To study the effect of variation of sampling frequency on the demodulated output.

APPARATUS:
1. Sampling and reconstruction Trainer.
2. C.R.O(30Mhz)
3. Patch cords.

CIRCUIT DIAGRAM:



KIT DIAGRAM



THEORY:

Pulse Modulation is used to Transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with synchronizing signals. At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

Sampling Theorem Statement:

A band limited signal of finite energy which has no frequency components higher than fm Hz, is completely described by specifying the values of the signal at instants of time separated by ½ fm seconds.

The sampling theorem states that, if the sampling rate in any pulse modulation system exceeds twice the maximum signal frequency, the original signal can be reconstructed in the receiver with minimum distortion.

Fs > 2fm is called Nyquist rate.

Where fs – sampling frequency

Fm – Modulation signal frequency.

If we reduce the sampling frequency fs less than fm, the side bands and the information signal will overlap and we cannot recover the information signal simply by low pass filter. This phenomenon is called fold over distortion or aliasing.There are two methods of sampling. (1) Natural sampling (2) Flat top sampling.

Sample & Hold circuit holds the sample value until the next sample is taken.

Sample & Hold technique is used to maintain reasonable pulse energy.The duty cycle of a signal is defined as the ratio of Pulse duration to the Pulse repetition period. The duty cycle of 50% is desirous taking the efficiency into account.

Pulse and Modulating Signal Generator:-

A 4.096 MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comprising a 4.096MHz crystal and three 74HC04(U9) inverter gates. This 4.096MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U10), to produce 50% duty cycle, 1KHz square wave on pin no.1 of U10, and 2KHz square wave on pin no.15. the frequency is selectable by means of SW1. this input of fourth order low pass filter U11(TL072) is used to produce sine wave from the square wave. The amplitude of this sine wave can be varied.

The square wave which is generated by the oscillator is buffered by inverter 74HC04(U9), to produce 32KHz square wave at pin no. 4 of the 74HC4040. This pulse is given to the monostable multi(U4) to obtain the 16KHz and 32KHz square wave at the output which are selected by the frequency pot.

Sampling Circuit:-

The IC DG211(U3) is used as analog switch which is used in pulse amplitude modulation in this circuit. The modulation signal & pulse signal are given as the input to TL074(U2), 7400(U1) IC‟s respectively. These IC output are fed to the inputs of the DG211.

The sampled output is available at the pin no.2 of DG211 and it is buffered by using TL074(U2) and then output is available at TP5.

Similarly the sample & hold output and the flat top output are available at pin no15 & 10 of DG211 respectively. These are buffered by TL074(U2) and then output is available at TP6 & TP7 respectively.

Reconstruction Circuit:-

The demodulation section comprises of a fourth order low pass filter and an AC amplifier. The TL074(U5) is used as a low pass filter and AC amplifier. The output of the modulator is given as the input to the low pass filter.
The low pass filter output is obviously less and it is fed to the AC amplifier which comprises of a single op amp and whose output is amplified.

PROCEDURE:

1. Connect the adaptor to the mains.

2. Observe the AF output in the CRO. It is a sine wave of frequency varying from 200Hz to 2KHz with0.5 Vp-p amplitude variations. Adjust the frequency to nearly 1KHz, with the help of potentiometer p1.

3. . Observe the output of the 8KHz sync clock generators. Its frequency can be varied using 100k trim pot. Adjust its free running frequency such that sync clock output is almost around 8KHz.

4. Connect the output of the AF signal generator to the sync signal input of the sync clk generator. Observe that both the signals are synchronised on the oscilloscope.

5. Connect the AF output of AF signal generator to the AF input of ASSR modulators.

6. Connect the AF signal input in CH1 & synchronous clk in CH2 of the oscilloscope. Choose the trigger at CH1. Both the signals will appear synchronized.

7. Now observe ASSR pulses with natural top sampling i.e the top of the pulse will follow the input signal during the sampling period. The output is given to the demodulators and demodulated output can be seen as replica of the AF output.

8. The effect of variation in AF frequency and amplitude can be studied at the output.

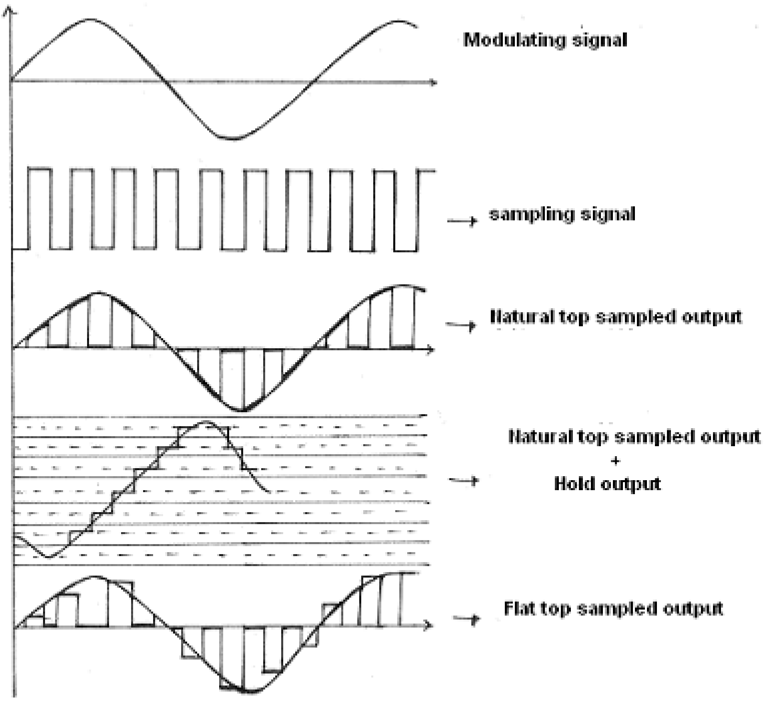
9. The sampled output will be a stepped sinewave whose amplitude is held constant for the rest of the period after the sampling pulse.

10. This is also a form of ASSR which can be demodulated & seen at the demodulated output.

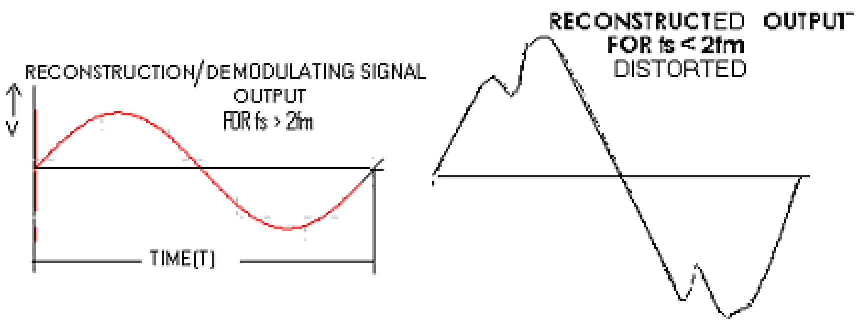
11. Now connect the CH2 of the oscillator to plat top output. We observe flat top ASSR pulse which are in the post sampling period.

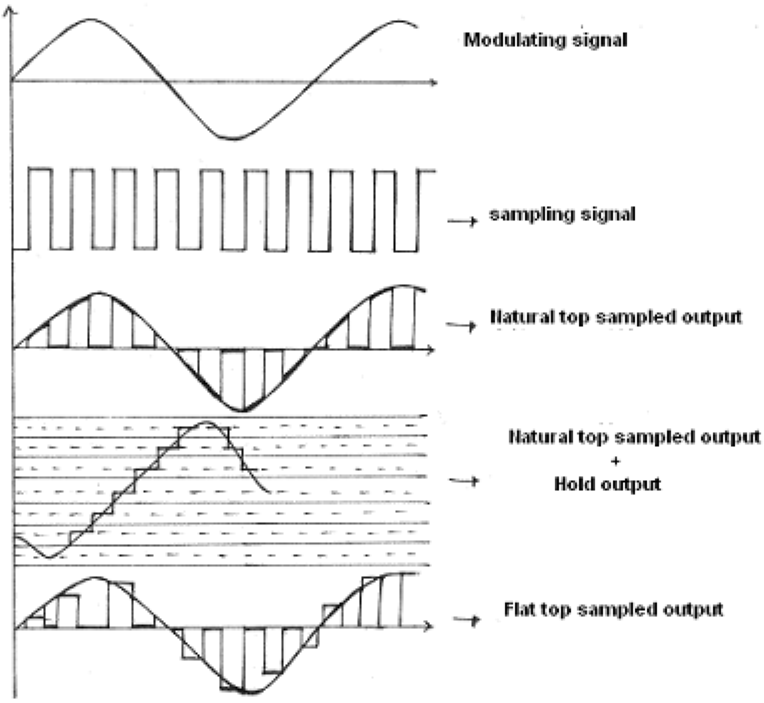
12. Connect these pulses to the demodulator & observe the demodulated output

EXPECTED WAVEFORMS

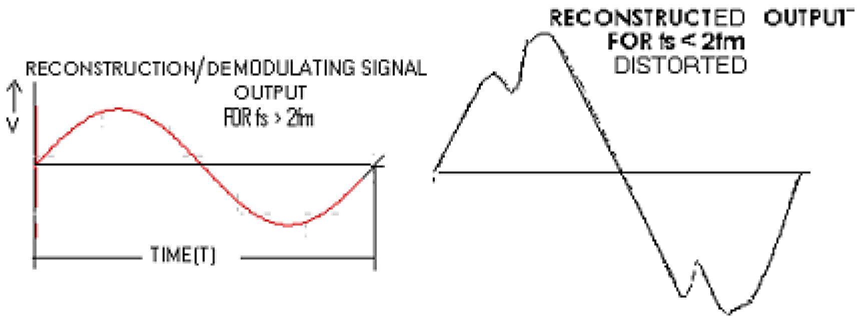


DEMODULATED OUTPUT





DEMODULATED OUTPUT



RESULT:

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QUESTIONS

1.What are the types of sampling?
2.State sampling theorem?
3.What happens when fs < 2 fm ?
4.How will be the reconstructed signal when fs >= 2fm?
5.Explain the operation of sampling circuit?
6. Explain the operation of re-construction circuit?
7.Who formalised the sampling theorem ?
8.What are the applications of the above theorem?
9.Is the sampling theorem basis for the modern digital communications?

MATLAB CODE:

```
tfinal=0.05;
t=0:0.0005:tfinal;
fm=input('enter the analog frequency');
xt=cos(2*pi*fm*t);
fs1=1.3*fm;
n1=0:1/fs1:tfinal;
xn=cos(2*pi*n1*fm);
subplot(3,1,1);
plot(t,xt,'b',n1,xn,'r');
title('under sampling plot');
fs2=2*fm;
n2=0:1/fs2:tfinal;
xn=cos(2*pi*fm*n2);
subplot(3,1,2);
plot(t,xt,'b',n2,xn,'r');
title('Nyquist plot');
fs3=5*fm;
n3=0:1/fs3:tfinal;
xn=cos(2*pi*fm*n3);
subplot(3,1,3);
plot(t,xt,'b',n3,xn,'r');
title('over sampling plot');
xlabel('time');
ylabel('amplitude');
legend('analog','discrete');
```

Input: Enter the analog frequency :100

SIMULATED OUTPUT WAVEFORMS

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ATTACH GRAPH SHEET HERE

EXPERIMENT NO-9

PULSE AMPLITUDE MODULATION

AIM:- 1.To study the Pulse amplitude modulation & demodulation Techniques.
2.To study the effect of amplitude and frequency variation of modulating signal on the output.

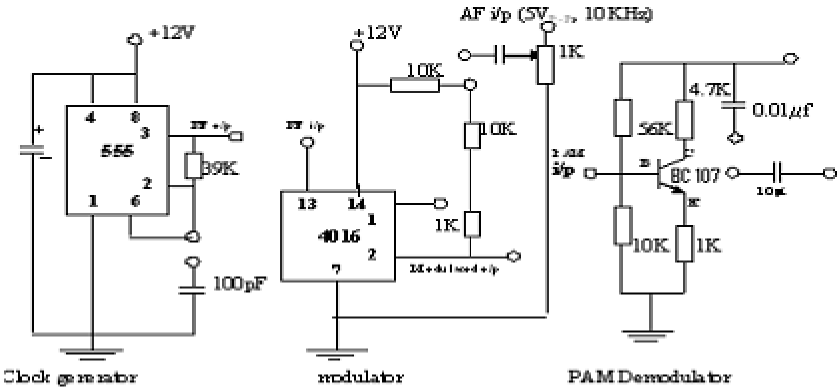
APPARATUS:-

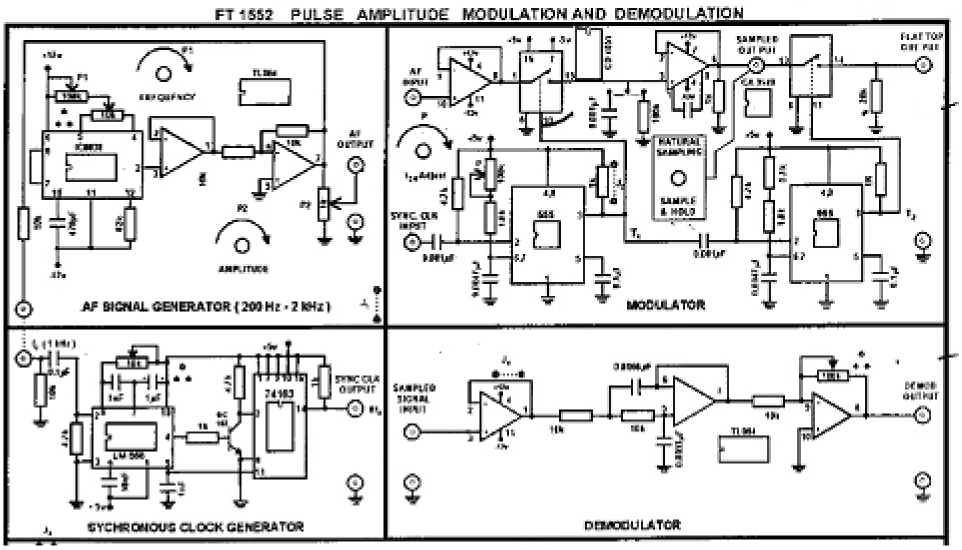
1. Pulse amplitude modulation & demodulation Trainer.

2. Dual trace CRO.

3. Patch chords.

CIRCUIT DIAGRAM:





THEORY:-

Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with syncing signals.At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The pulse amplitude modulation is the simplest form of the pulse modulation. PAM is a pulse modulation system is which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cables are used to modulated carrier.The two types of PAM are i) Double polarity PAM, and ii) the single polarity PAM, in which a fixed dc level is added to the signal to ensure that the pulses are always positive. Instantaneous PAM sampling occurs if the pulses used in the modulator are infinitely short.

Natural PAM sampling occurs when finite-width pulses are used in the modulator, but the tops of the pulses are forced to follow the modulating waveform.Flat-topped sampling is a system quite often used because of the ease of generating the modulated wave.PAM signals are very rarely used for transmission purposes directly. The reason for this lies in the fact that the modulating information is contained in the amplitude factor of the pulses, which can be easily distorted during transmission by noise, crosstalk, other forms of distortion. They are used frequently as an intermediate step in other pulse-modulating methods, especially where time-division multiplexing is used.

PROCEDURE:

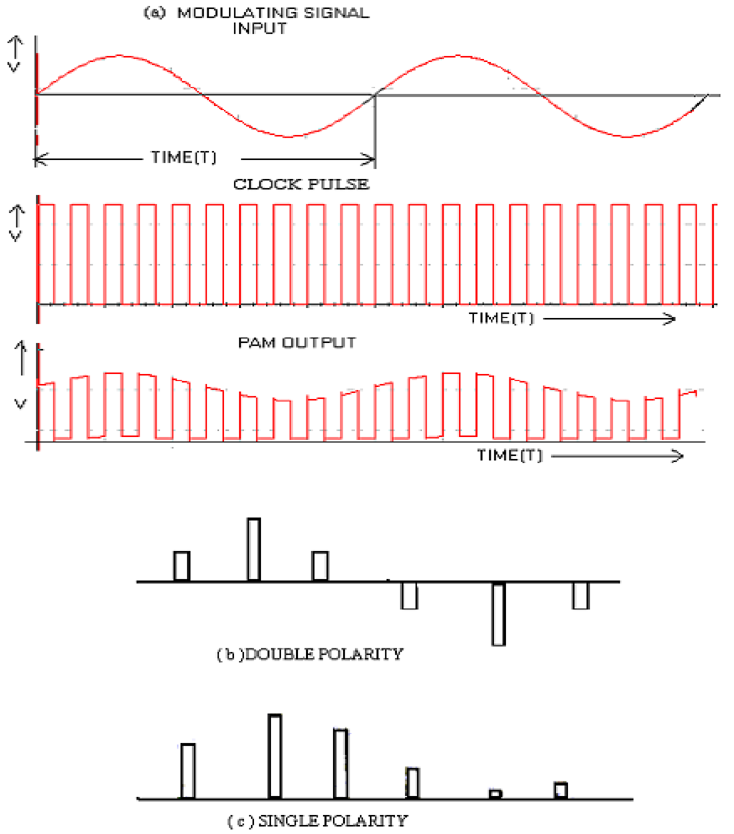
1. Connect the adaptor to the mains.
2. Observe the AF output in the CRO. It is a sine wave of frequency varying from 200Hz to 2KHz with0.5 Vp-p amplitude variations. Adjust the frequency to nearly 1KHz, with the help of potentiometer p1.
3. . Observe the output of the 8KHz sync clock generators. Its frequency can be varied using 100k trim pot. Adjust its free running frequency such that sync clock output is almost around 8KHz
4. Connect the output of the AF signal generator to the sync signal input of the sync clk generator showed at dashed at dashed line. Obsreve both the signals are synchronized on the oscilloscope. For this purpose feed the AF signal to CH1 of the oscilloscope & the sync clk output to CH2of the oscilloscope. Select the trigger sources as CH1 & observe that both the signals are locked. Otherwise slowly adjust the AF signal frequency on either side of 1KHz to get the frequency clock.
5. Connect the AF output of AF signal generator to the AF input of PAM modulator.

6. Connect the AF signal input in CH1 & synchronous clk in CH2 of the oscilloscope. Choose the trigger at CH1. Both the signals will appear synchronized.

7. Keep the switch in natural sampling mode. Connect the CH2 of the oscilloscope to observe the sampled output.
8. Now observe PAM pulses with natural top sampling i.e top of the pulses will follow the input signal during the sampling period. This output can be given to demodulator & the demodulated output can be seen as replica of the AF output.
9. The effect of variation in AF frequency & amplitude can be studied at the output.
10. Alternating the frequency & amplitude of the AF signal is brought back to 1KHz. Now the switch is kept in sample &hold mode.

11. The sampled output will appear & stepped sine wave whose amplitude is held constant for the rest of the period after the sampling pulse.
12. This is also o form of PAM which can be demodulated & seen at the demodulated output.
13. Now connect the CH2 of the oscilloscope to flat top output. We observe flat top PAM pulses which are present in the post sampling period.
14. Connect the pulses to the demodulator and observe the demodulated output.
15. Thus the trainer is useful in observing natural sampled PAM pulses, S & H PAM pulses & flat top PAM pulses at different frequencies and amplitudes of the AF signal.

EXPECTED WAVEFORMS



RESULT:

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QUESTIONS

1.TDM is possible for sampled signals. What kind of multiplexing can be used in continuous modulation systems?
2.What is the minimum rate at which a speech signal can be sampled for the purpose of PAM?
3.What is cross talk in the context of time division multiplexing?

MATLAB CODE

```
clc;
clf;
close all
clear all
t=0:1/6000:((10/1000)-(1/6000));
xa=sin(2*pi*100*abs(t));
Ts=32;
x=sin(2*pi*600*(Ts*t));
X=fft(xa,abs(x));
subplot(3,1,1)
plot(xa);
grid
subplot(3,1,2);
stem(X);
grid
Y=ifft(xa,X);
subplot(3,1,3)
plot(Y)
grid
```

SIMULATED OUTPUT

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EXPERIMENT NO-10

PULSE WIDTH MODULATION & DEMODULATION

AIM:

1.To study the Pulse Width Modulation (PWM) and Demodulation Techniques.

2. To study the effect of Amplitude and Frequency of Modulating Signal on PWM output.

APPARATUS:

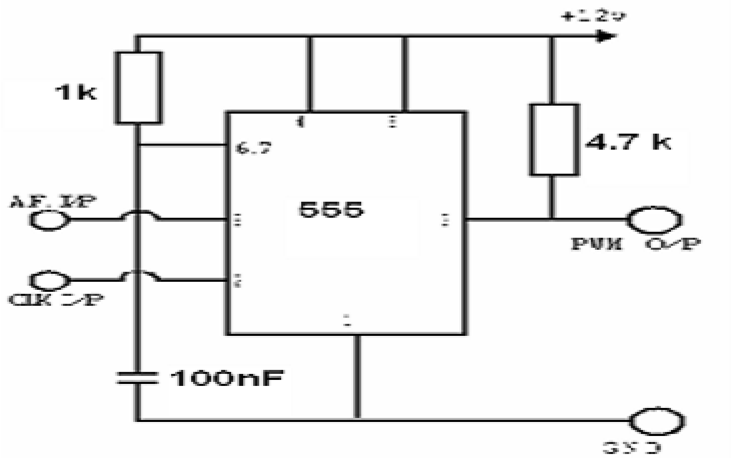
1. PWM trainer kit

2. C.R.O(20MHz)

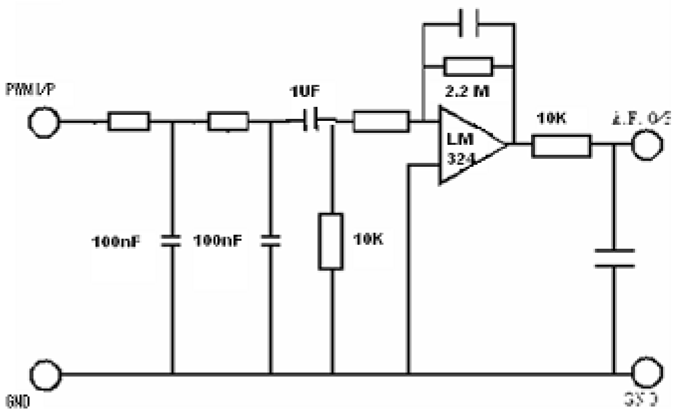
3. Patch Chords.

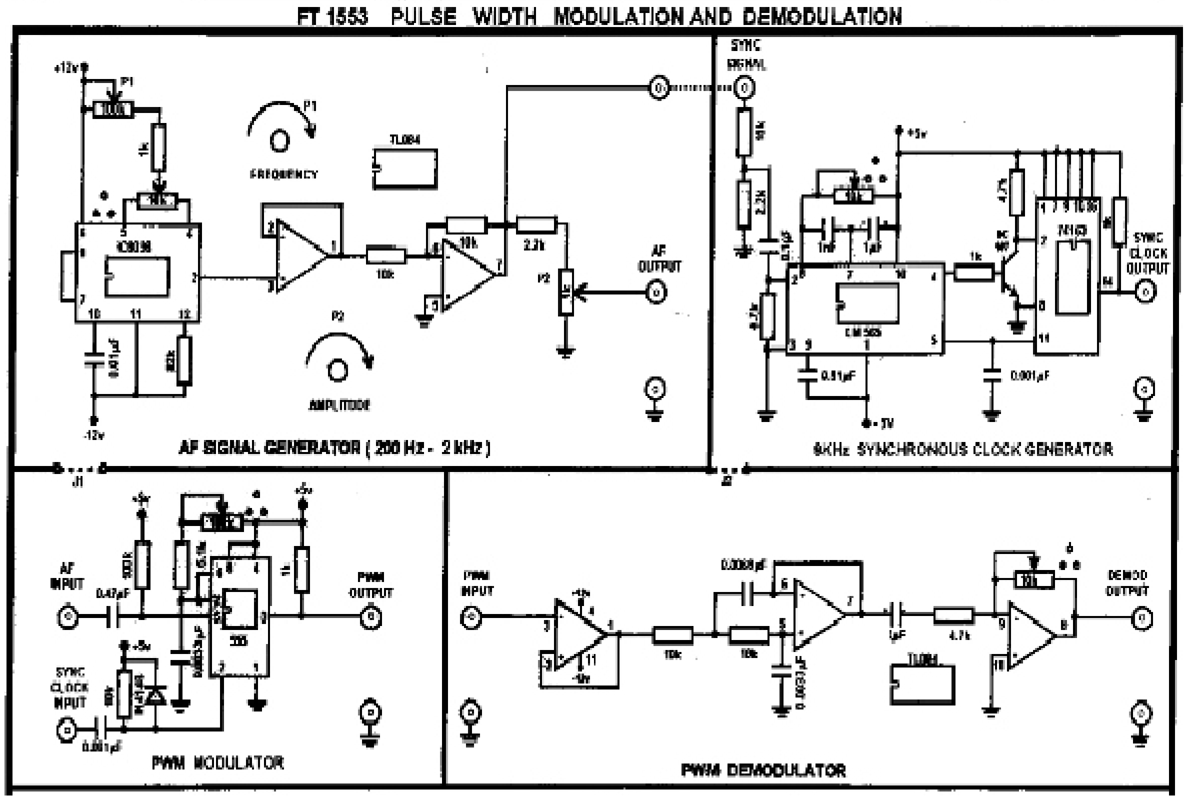
CIRCUIT DIAGRAM:

PWM Modulator



PWM Demodulator





THEORY:-

Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with synchronizing signals.At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The pulse Width Modulation of the PTM is also called as the Pulse Duration Modulation(PDM) & less often Pulse length Modulation(PLM).In pulse Width Modulation method, we have fixed and starting time of each pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant.This method converts an amplitude varying message signal into a square wave with constant amplitude and frequency, but which changes duty cycle to correspond to the strength of the message signal.

Pulse-Width modulation has the disadvantage, that its pulses are of varying width and therefore of varying power content. This means that the transmitter must be powerful enough to handle the maximum-width pulses. But PWM still works if synchronization between transmitter and receiver fails, whereas pulse-position modulation does not.Pulse-Width modulation may be generated by applying trigger pulses to control the starting time of pulses from a monostable multivibrator, and feeding in the signal to be sampled to control the duration of these pulses. When the PWM signals arrive at its destination, the recovery circuit used to decode the original signal is a sample integrator(LPF).

PROCEDURE:

1. Connect the adaptor to the mains.

2. Observe the AF output in the CRO. It is a sine wave of frequency varying from 200Hz to 2KHz with0.5 Vp-p amplitude variations. Adjust the frequency to nearly 1KHz, with the help of potentiometer p1.

3. . Observe the output of the 8KHz sync clock generators. Its frequency can be varied using 100k trim pot. Adjust its free running frequency such that sync clock output is almost around 8KHz

4. Connect the output of the AF signal generator to the sync signal input of the sync clk generator showed at dashed at dashed line. Obsreve both the signals are synchronized on the oscilloscope. For this purpose feed the AF signal to CH1 of the oscilloscope & the sync clk output to CH2of the oscilloscope. Select the trigger sources as CH1 & observe that both the signals are locked. Otherwise slowly adjust the AF signal frequency on either side of 1KHz to get the frequency clock.

5. Connect the AF output of AF signal generator to the AF input of PWM modulator. Connect the sync clock output to the sync clock input of PWM.

6. Connect the AF signal input in CH1& sync clock in CH2 of the oscilloscope. Choose the trigger to CH1. Both the signals will appear synchronized.

7. Observe the sync clk input in CH1 & PWM output to CH2 of CRO. Selct the trigger source to CH1 & trigger edge to –ve.

8. When the AF signal amplitude is zero, the PWM pulses are observed as varying +ve pulses. All the pulses start at –ve edge of clock & end as per the instantaneous amplitude of AF signal.

9. Now increase the amplitude of the AF signal. The pulse width varies on right side of rising edge marked. At reference the variation in tune corresponds to amplitude of input. At that instant measure the amplitude & time variation which gives a correspondence of μs/v modulation quotient.

10. Connect the PWM output of modulator to PWM input of the demodulator. Observe the demodulated output which is same as the input.

11. Connect the AF output on CH1 & demodulated output on CH2, vary the AF output amplitude & observe the corresponding variations in the output of demodulation

EXPECTED WAVEFORMS

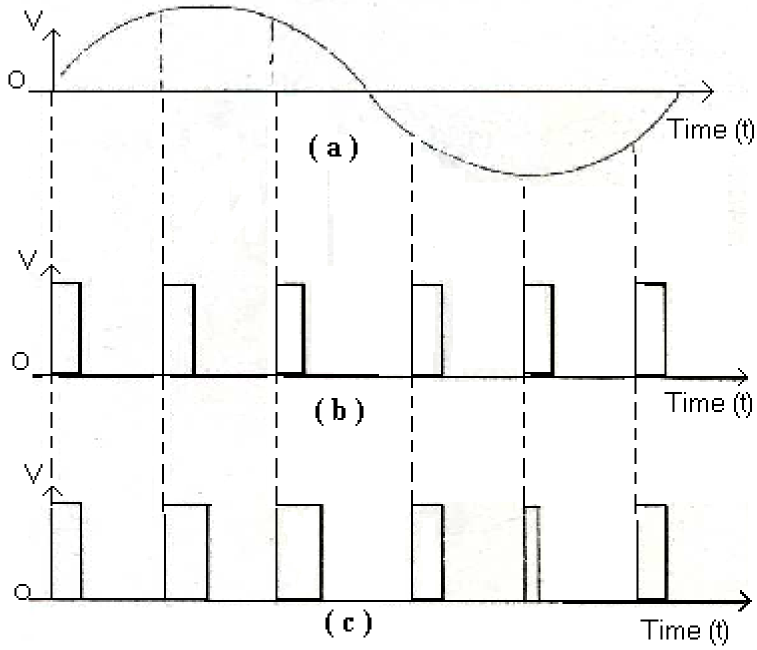


Fig (2) PULSE WIDTH MODULATION
(a) Signal
(b) Unmodulated pulses
(c) PWM

RESULT:

QUESTIONS

1.An audio signal consists of frequencies in the range of 100Hz to 5.5KHz.What is the minimum frequency at which it should be sampled in order to transmit it through pulse modulation?
2.Draw a TDM signal which is handling three different signals using PWM?
3.What do you infer from the frequency spectrum of a PWM signal?
4.Clock frequency in a PWM system is 2.5kHz and modulating signal frequency is 500Hzhowmany pulses per cycle of signal occur in PWM output? Draw the PWM signal?
5.Why should the curve for pulse width Vs modulating voltage be linear?
6.What is the other name for PWM?
7.What is the disadvantage of PWM?
8.Will PWM work if the synchronization between Tx and Rx fails?
9.Why integrator is required in demodulation of PWM?

MATLAB CODE:

```
fc=1000;
fs=10000;
f1=200;
t=0:1/fs:((2/f1)-(1/fs));
x1=0.4*cos(2*pi*f1*t)+0.5;
%modulation
y1=modulate(x1,fc,fs,'pwm');
subplot(421);
plot(x1);
title('original signal tokne mesage,f1=500,fs=10000')
subplot(422);
plot(y1);
axis([0 500 -0.2 1.2]);
title('PWM')
%demodulation
x1_recov=demod(y1,fc,fs,'pwm');
subplot(423);
plot(x1_recov);
title('time domain reoverd recoverd,sigle tone,f1=200')
```

SIMULATED OUTPUT

RESULT

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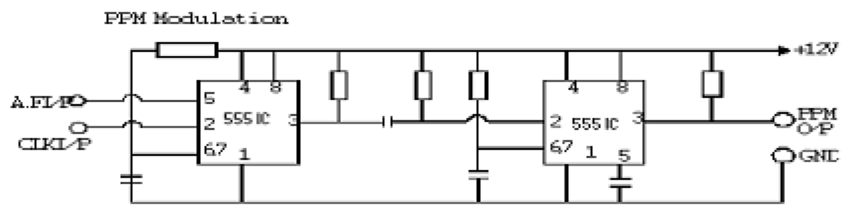
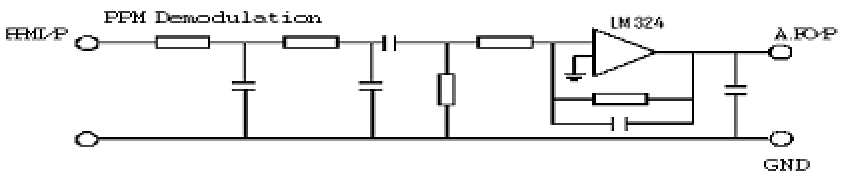
EXPERIMENT NO-11
PULSE POSITION MODULATION AND DEMODULATION

AIM:
1. To study the generation Pulse Position Modulation (PPM) and Demodulation.
2. To study the effect of Amplitude and the frequency of modulating signal on Its output and observe the wave forms.

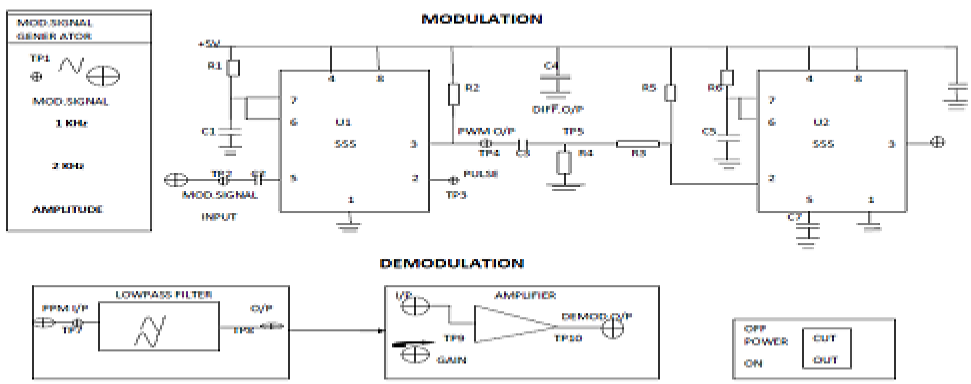
APPARATUS:
1. Pulse Position Modulation (PPM) and demodulation Trainer.
2. C.R.O(20MHz)
3. Patch chords.

CIRCUIT DIAGRAM:





(OR)



THEORY:-

Pulse Modulation is used to transmit analog information in this system continuous wave forms are sampled at regular intervals. information regarding the signal is transmitted only at the sampling times together with synchronizing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples. Pulse modulation may be subdivided in to two types analog and digital. In analog the indication of sample amplitude is the nearest variable. In digital the information is a code.

The pulse position modulation is one of the methods of the pulse time modulation.PPM is generated by changing the position of a fixed time slot.

The amplitude& width of the pulses is kept constant, while the position of each pulse, in relation to the position of the recurrent reference pulse is valid by each instances sampled value of the modulating wave. Pulse position modulation into the category of analog communication. Pulse-Position modulation has the advantage of requiring constant transmitter power output, but the disadvantage of depending on transmitter receiver synchronization.

Pulse-position modulation may be obtained very simply from PWM. However, in

PWM the locations of the leading edges are fixed, whereas those of the trailing edges are not. Their position depends on pulse width, which is determined by the signal amplitude at that instant. Thus, it may be said that the trailing edges of PWM pulses are, in fact, position-modulated. This has positive-going narrow pulses corresponding to leading edges and negative-going pulses corresponding to trailing edges. If the position corresponding to the trailing edge of an un modulated pulse is counted as zero displacement, then the other trailing edges will arrive earlier or later. They will therefore have a time displacement other than zero; this time displacement is proportional to the instantaneous value of the signal voltage. The differentiated pulses corresponding to the leading edges are removed with a diode clipper or rectifier, and the remaining pulses, are position-modulated.

PROCEDURE:

1. Connect the adaptor to the mains.

2. Observe the AF output in the CRO. It is a sine wave of frequency varying from 200Hz to 2KHz with0.5 Vp-p amplitude variations. Adjust the frequency to nearly 1KHz, with the help of potentiometer p1.

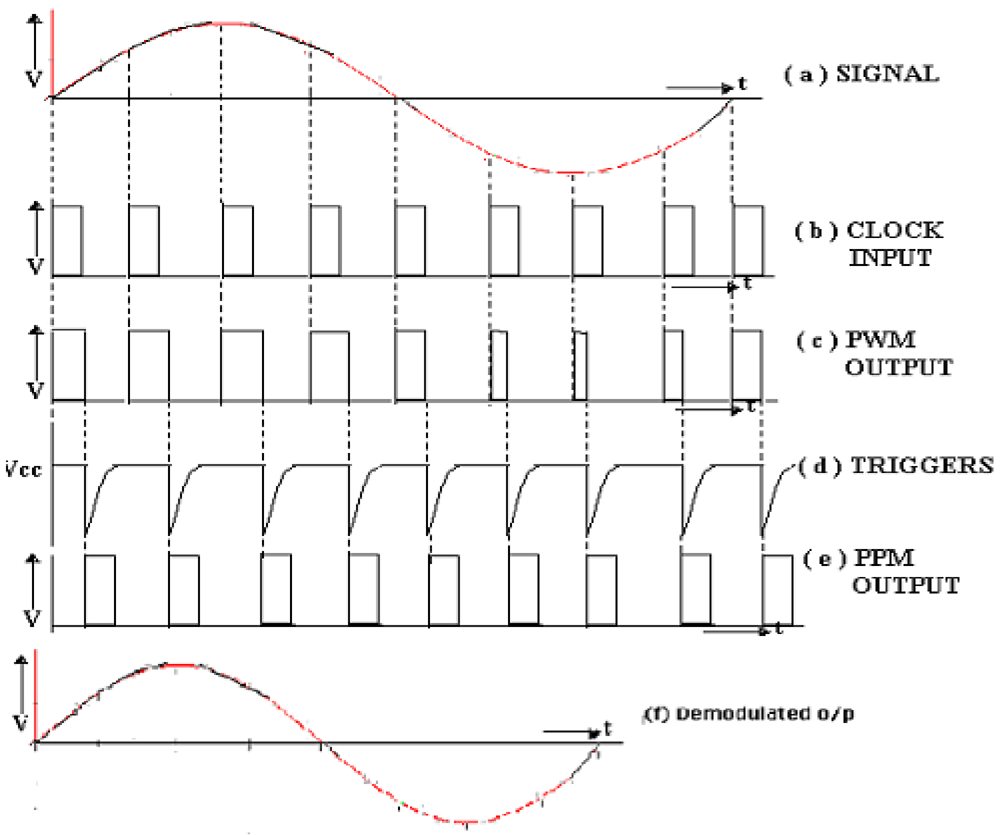
3. . Observe the output of the 8KHz sync clock generators. Its frequency can be varied using 100k trim pot. Adjust its free running frequency such that sync clock output is almost around 8KHz

4. Connect the output of the AF signal generator to the sync signal input of the sync signal input of the sync clock generator. Obsreve both the signals are synchronized on the oscilloscope.

5. For this purpose feed the AF signal to CH1 of the oscilloscope & the sync clk output to CH2of the oscilloscope. Select the trigger sources as CH1 & observe that both the signals are locked. Otherwise slowly adjust the AF signal frequency on either side of 1KHz to get the frequency clock.

6. Connect the AF output of AF signal generator to the AF input of PPM modulator. Connect the sync clock output to the sync clock input of PPM.
7. Connect the AF signal input in CH1& sync clock in CH2 of the oscilloscope. Choose the trigger to CH1. Both the signals will appear synchronized.
8. When the AF signal amplitude is zero, the PPM pulses are observed as narrow pulses spaced equidistant & at constant time delay w.r.t –ve edge of the clk.
9. Now increase the amplitude of the AF signal. The pulse position varies on either side of the reference position marked at reference earlier.
10. Adjust the AF amplitude to zero using pot P1 using the uncalibrated knob of the oscilloscope adjust the waveform such that each pulse is at the beginning of each division of the scope time scale.
11. At this stage each division is equal to 7ms there are now 10 pulses on the screen just more than 1cycle of sinewave.
12. Now increase the amplitude of the sine wave to max. Connect the CH1 of the sinewave to max. Connect the CH1 of the CRO to AF input. Note the displacement of each of the pulses on CH2 from the original position & the corresponding amplitude of AF signal at the beginning of the division.
13. Connect the PPM output of modulator to PPM input of the demodulator. Observe the demodulated output which is same as the input.
14. The sync clk generator will not synchronise for all range of frequencies of the AF signal generator.

EXPECTED WAVEFORMS:



RESULT:

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QUESTIONS:

1.What is the advantage of PPM over PWM?
2.Is the synchronization is must between Tx and Rx
3.Shift in the position of each pulse of PPM depends on what?
4.Can we generate PWM from PPM?
5.Why do we need 555 timer?
6.Does PPM contain derivative of modulating signal compared to PWM?
7.For above scheme, do we have to use LPF and integrator in that order.?
8.If we convert PPM to PWM & then detect the message signal, will the o/p has less distortion?
9.Is synchronization critical in PPM ?
10.How robust is the PPM to noise ?

MATLABCODE

```
clc;
clear all;
close all;
fc=100;
fs=1000;
f1=80;%f2=300
t=0:1/fs:((2/f1)-(1/fs));
x1=0.4*cos(2*pi*f1*t)+0.5;
%x2=0.2*(cos(2*pi*f1*t)+cos(2*pi*f2*t))+0.5 ;
subplot(4,2,1)
plot(x1)
title('original msg signal')
y1=modulate(x1,fc,fs,'ppm')
subplot(4,2,2)
plot(y1)
axis([0 50 -0.2 1.2])
title('ppm one of f1,fc=1000,f1=80 ')
fx1=abs(fft(y1,1024))
fx1=[fx1(512:1024) fx1(1:513)]
f=[(511*fs/1024):(fs/1024):(512*fs/1024)]
subplot(4,2,3)
plot(fx1)
title('freq des ppm signal tone,fc=1000')
x1_recov = demod(y1,fc,fs,'ppm')
subplot(4,2,4)
plot(x1_recov)
title('time domain recovered signal')
```

STIMULATED OUTPUT WAVEFORMS

SIGNATURE OF LAB IN-CHARGE

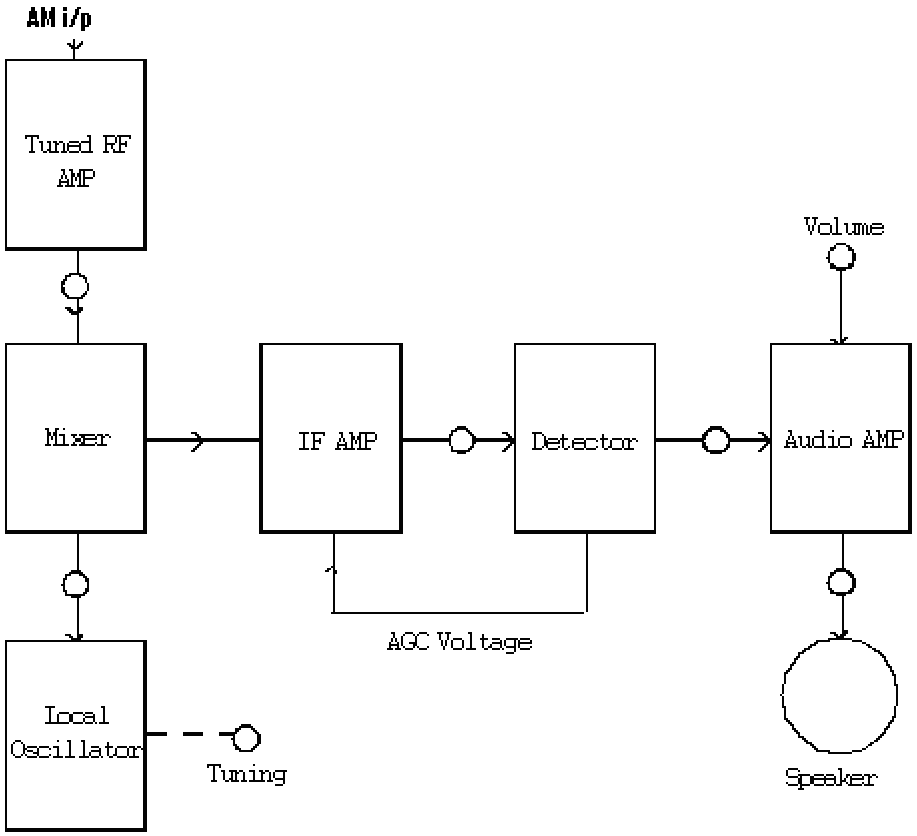
ATTACH ONE GRAPH SHEET HERE

EXPERIMENT NO-12
AGC CHARACTERISTICS

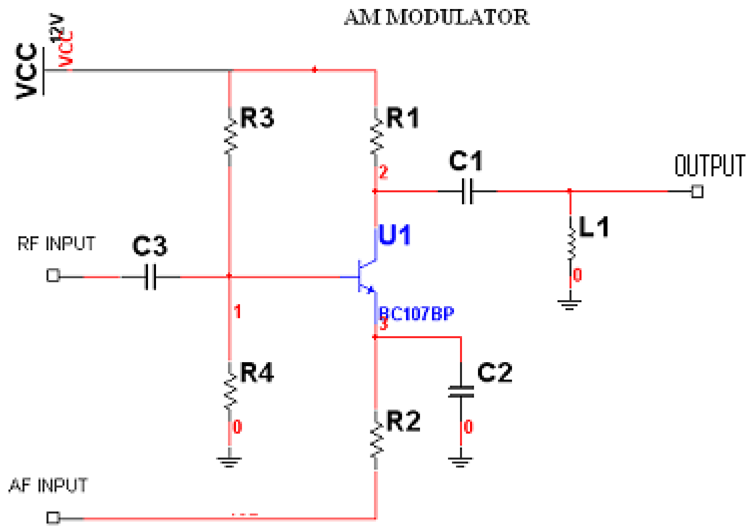
AIM:- To study the operation of AGC in a Communication system

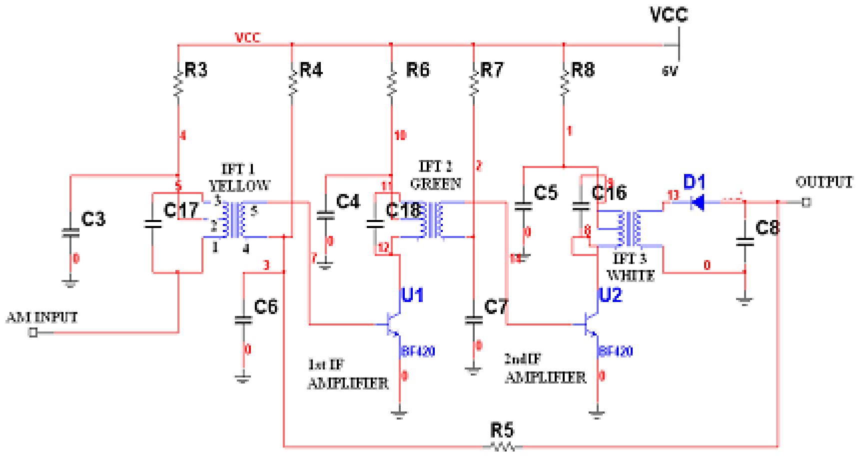
APPARATUS:
1.AGC Trainer kit
2.Dual Trace Oscilloscope
3.Digital Multimeter.

BLOCK DIAGRAM:



CIRCUIT DIAGRAM:





THEORY:

A simple AGC is a system by means of which the overall gain of a radio receiver is varied automatically with the changing strength of the received signal, to the out put substantially constant. A DC bias voltage, derived from the detector. The devices used in those stages are ones who `s trans conductance and hence gain depends on the applied bias voltage or current. It may be noted in passing that, for correct AGC operation, this relationship between applied bias and trans-conductance need not to be strictly linear, as long as trans-conductance drops significantly with increased bias. All modern receivers are furnished with AGC, which enables tuning to stations of verifying signal strengths without appreciable change in the size of the output signal thus AGC "irons out" input signal amplitude variations, and the gain control does not have to be readjusted every time the receiver is tuned from one station to another, expect when the change in signal strengths is enormous. In addition, AGC helps

To smooth out the rapid fading which may occur with long distance short wave reception and prevents the overloading of the last IF amplifier which might otherwise have occurred.

PROCEDURE:

1.As the circuit is already wired you just have to trace the circuit according to the circuit diagram given above Fig1.1

2.Connect the trainer to the mains and switch on the power supply.

3.Measures the output voltages of the regulated supply i.e. +12v and -12v, +6@150ma.

4.Observe outputs of RF and AF signal generator using CRO, note that RF voltage is approximately 50mVpp of 455Khz frequency and AF voltage is 5Vpp of 1Khz frequency.

5.Now vary the amplitude of AF signal and observe AM wave at output, note the percentage of modulation for different values of AF signal.

% Modulation= (B-A)/(B+A) *100

6.Now adjust the modulation index to 30% by varying the amplitudes of RF & AF signals simultaneously.

7.Connect AM output to the input of AGC and also to the CRO channel-1.

8.Connect AGC link to the feedback network through OA79 diode

9.Now connect CRO channel-2 at output. The detected audio signal of 1Khz will be observed.

10.Calculate the voltage gain by measuring the amplitude of output signal (Vo)

Wave form using formula A =Vo/Vi.

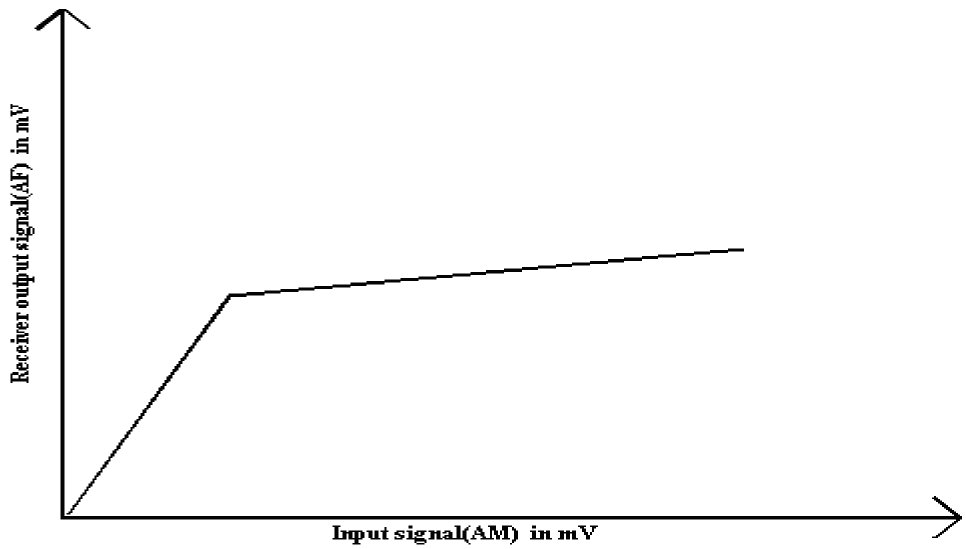
11.Now vary input level of 455Khz IF signal and observe detected 1Khz audio signal with and without AGC link. The output will be distorted when AGC link removed i.e. there is no action.

12.This explains AGC effect in Radio circuit.

TABULAR COLUMN:

S.NO	RF INPUT VOLTAGE (X-axis)	AF out put Voltage (Y-axis)

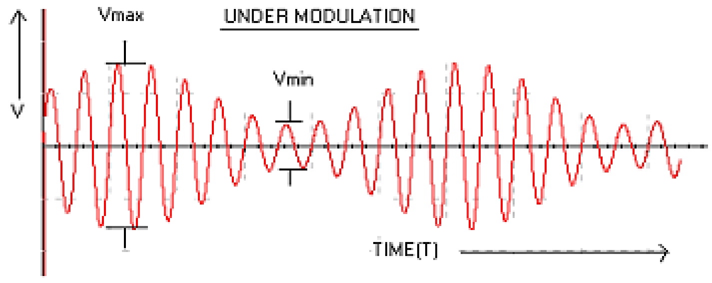
EXPECTED GRAPH:



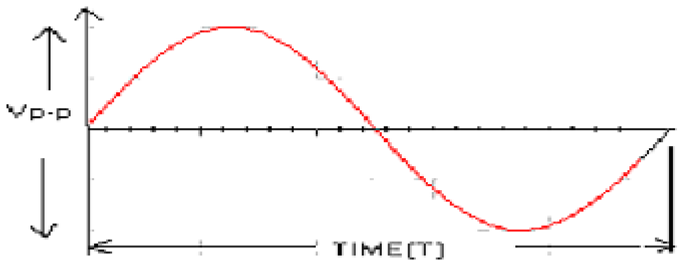
AGC characteristics curve

EXPECTED WAVE FORMS:

AM INPUT



DETECTED OUTPUT WITH AGC



RESULT:

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MATLAB CODE:

```
%AGC characteristics
clc;
clear;
t0=0.15; %signal duration
ts=1/4000; %sampling intervel
fc=250; %carrier frquency
fs=1/ts; %sampling frequency
df=0.25; %desired fre.resolution
t=(0:ts:t0-ts); %time vector
%modulating signal
m =[ones(1,t0/(3*ts)),ones(1,t0/(3*ts)),ones(1,t0/(3*ts))];
%carrier signal
c =cos(2*pi*fc*t);
%modulated signal
mod_signal= m.*c;
rate=fs/fc;
framelength=length(mod_signal)/rate;
%calculation of analytic signal using hilbert transform
analytic_signal=hilbert(mod_signal);
l=real(analytic_signal); %real part of analytic signal
Q=imag(analytic_signal); %imaginary part of analytic signal
%scaling the modulated signal by different amplitudees and adding random
%noise
%so as to analyze the performance of AGC
idecision=[l(1:length(l)/2)*4
l(length(l)/2+1:length(l))*10]+0.1*randn(1,rate*framelength);
qdecision=[Q(1:length(Q)/2)*4
Q(length(Q)/2+1:length(Q))*10]+0.1*randn(1,rate*framelength);
threshold=20; %threshold value setting for AGC
upperlimit=15; %setting the upperlimit for maximum amplitude signal
lowerlimit=0; %setting the lowerlimit for minimum amplitude signal
%parameter initialization for performing AGC
sum(1)=1;
limited(1)=1;
gain(1)=1;
iscaled(1)=idecision(1);
qscaled(1)=idecision(1);
iabs(1)=abs(iscaled(1));
qabs(1)=abs(qscaled(1));
hi_lo(1)=0;
```

```
error(1)=0;

%AGC loop
for i=2:rate*framelength
%performing gain control using the gain value calculated in the
%previous iteration
iscaled(i)=gain(i-1)*idecision(i);
qscaled(i)=gain(i-1)*qdecision(i);
%performence relative amplitude scaling between l and Q
iabs(i)=abs(iscaled(i));
qabs(i)=abs(qscaled(i));
if iabs(i)>qabs(i)
amplitude(i)=iabs(i)+0.5*qabs(i);
else
amplitude(i)=qabs(i)+0.5*iabs(i);
end;
%calculation of gain values using the threshold as well as the upper
%and lower limit setting
if amplitude(i)>threshold
error(i)=0.05;
else
error(i)=0.05;
end;
sum(i)=limited(i-1)+error(i);
if sum(i)>upperlimit;
limited(i)=upperlimit;
elseif sum(i)<lowerlimit
limited(i)=lowerlimit;
else
limited(i)=sum(i);
end;
gain(i)=limited(i-1);
end
figure(1);
subplot(3,1,1);
plot(idecision);
title('original l signal');
subplot(3,1,2);
plot(iscaled);
title('l signal after the gain control');
subplot(3,1,3);
```

```
plot(gain);
title('gain value');
figure(2);
subplot(3,1,1);

plot(qdecision);
title('original q signal');
subplot(3,1,2);
plot(qscaled);
title('q signal after the gain control');
subplot(3,1,3);
plot(gain);
title('gain value');
```

SIMULATED OUTPUT

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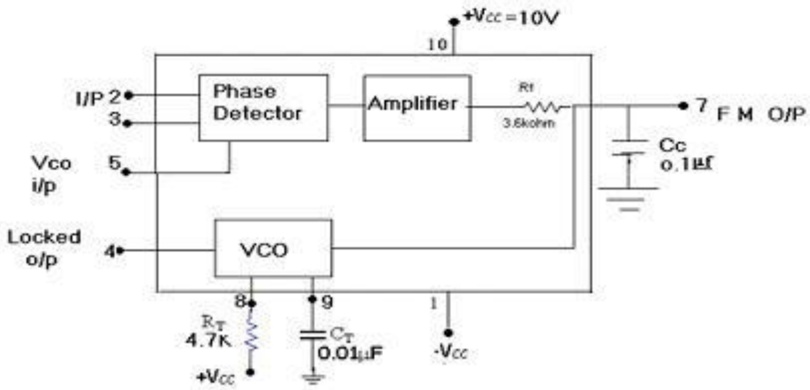
EXPERIMENT NO-13
PLL AS FM DEMODULATOR

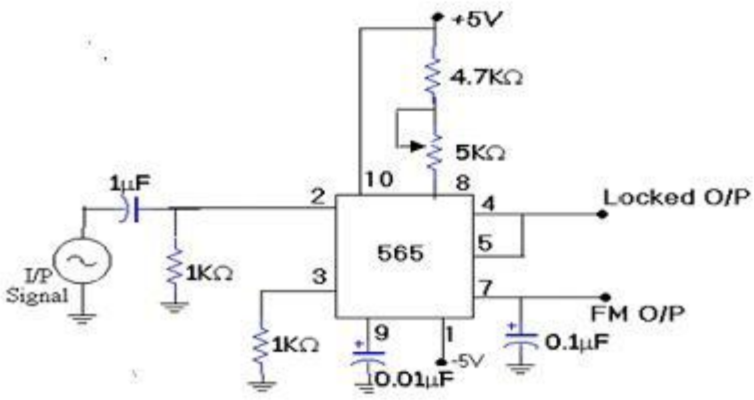
AIM: Perform the characteristics of PLL and also determine the Lock range and Capture range.

APPARATUS:
1. PLL trainer kit
2. CRO (20MHZ)
3. Digital Multimeter

PLL B LOCK DIAGRAM:



PLL CIRCUIT DIAGRAM:



THEORY:

Phase locked Loop is a versatile electronic servo system that compares the phase and frequency of a given signal with an internally generated reference signal. It is used in various applications like frequency multiplication,FM detector,AM modulator and Demodulator and FSK etc.,

Free running frequency (fo):

When there is no input signal applied to pin no:2 of PLL,it is free running frequency is determined by the circuit elements Rt and Ct and is given by
Fo= .3/(Rt Ct) where Rt is the timing resistor
Ct is the timing capacitor
Lock range of PLL (fl):
Lock range of PLL is in the range of frequencies in which PLL will remain lock, and this is given by
Fl= + 8fo/Vcc where fo is the free running frequency
Vcc = Vcc –(-Vcc)
= 2Vcc
Capture range (fc):
The capture range of PLL is the range of frequencies over which PLL acquires the lock. This is given by
Fc= 1/2pi underroot of 2pi fl /3.6x10cube xCc
Where fl is the lock range and Cc is filter capacitor
R = 3.6 * 10 cube
PROCEDURE:

1. Switch on the trainer and measure the output of the regulated supplies I.e, +12V and -5V
2. Observe the output of the square wave generator-using oscilloscope and measure the frequency range. The frequency range should be around 1Khz to 10Khz.
3. Calculate the free running frequency range of the circuit for different values of timing capacitor and Rt.
4. Connect 0.1micro Farad capacitor (Cc) to the circuit and open the loop by remoiving short between pin4&5. Measure the minimum and maximum free running frequencies obtainable at the output of the PLL (pin4) by varying the pot.
5.Compare your results with your results with your calculation from step 3 (theoretical value). Simultaneously you can observe the output signal using CRO.

Table 1:1 Free running frequency

Rt value (pot resistance in ohms)	Theoretical value (frequency in Khz)	Practical value (frequency in KHZ)

Lock range:

5. Calculate the lock range of the circuit for a 5Khz free running frequency and record in table 1.2

6. Connect pins 4,5 with the help of springs and adjust pot to get a free running frequency of 5Khz . Connect square wave generator output to the input of PLL circuit. Provide a 5Khz square signal of 1Vpp approximately (makes this input frequency as close to the Vcc frequency as possible).
7. Observe the input and output of the PLL.
8. Observe the input and output frequencies while slowly increasing the frequency of the square wave at the input. For same range output and input are equal (this is known as lock Range and PLL is said to be in lock with the input signal.) Record the frequencies at which the PLL breaks lock. (output frequency of PLL will be around Vco frequency and in oscilloscope you will see a jittery waveform when it breaks lock instead of clean square wave.) this frequency is called as upper end of the lock range and records this as F2.

9. Beginning at 5Khz , slowly decreases the frequency of the input and determine the frequency at which the PLL breaks lock on the low end record it as F1.

10. Find the lock range from F2-F1 and compare it with the theoretical values from step5.

Lock range table 1.2:

Theoretical value (frequency in Khz)	Practical value (frequency in Khz)

Capture range:

11. Calculate the capture range of the circuit for 5Khz free running frequency (consider filter capacitor (Cc)is 0.1mf).

12. With the oscilloscope and counter still on pin 4, slowly increase the input frequency from minimum (say 1Khz), Record the frequency (as F3) at which the input and output frequencies of the PLL are equal, this is known as lower end of the capture range.

13. Now keep the input frequency at maximum possible (say 10Khz), and slowly reduce the record frequency (as F4) at which the input and output frequencies of PLL are equal. This is known as upper end of the capture range.

14. Find capture range from F4-F3 and compare it with the theoretical value (from step 11).

15. Repeat the steps from 11 to 14 with Cc value 0.2Mf.

Capture range:

Filter Capacitor (Cc)	Theoretical value	Practical value

RESULT:

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QUESTIONS:

1. What are the applications of PLL?

2. What is a PLL ?

3. What is a VCO?

4. Define the lock range of PLL ?

5. Define the capture range of PLL ?

6. Give the expression for free running frequency fo of PLL?

7. What is meant by the free running frequency of a PLL?

8. Give the formulae for the lock range and capture range of the PLL?

MATLAB CODE

```
%PLL
clear all;
clc;
%definition of parameters
theta = pi/3; %Initial phase offset
f= 100; % CW frequency
fs = 100000; %sampling frequency
%Create the real and imaginaryparts of a CW nonmodulated carrier to be
%tracked
k = 1:1000;%define the no.of samples that needs to be generated
delf = f/40; %frequency deviation
%Generation of CW nonmodulated carrier
signal = exp(j*(2*pi*k*(f+delf)/fs + theta)); %+0.01*(rand(1,1000) + j*rand(1,1000));
%PLL Loop initialization
phi_hat(1) = 0;%phase estimate
e(1) = 0;%Error estimate
phd_output(1) = 0; %Phase detector output
nco(1) = 0;%Numerally controlled oscillator output
%PLL Filter parameters
kp = 0.15; %proportional constant
ki = 0.1; %Integer constant
% PLL Loop operation
for n=2:length(signal)
nco(n) = conj(exp(j*(2*pi*n*f/fs + phi_hat(n-1)))); % Compute nCO
phd_output(n) = imag(signal(n)*nco(n)); %Complex multiply nCO * input
e(n) = e(n-1) +(kp +ki)*phd_output(n)-ki*phd_output(n-1);%Filter integer
integ(n) = (kp +ki)*phd_output(n)-ki*phd_output(n-1);
phi_hat(n) = phi_hat(n-1)+ e(n);%update nCO
end
%Plotting the PLL waveforms at different stages
index_stop = 200;
figure(1);
plot(1:index_stop,phd_output(1:index_stop));
ylabel('Phase detector output')
figure(2);
plot(1:index_stop,phi_hat(1:index_stop)*180/pi,'m');
ylabel('Estimated phase');
index_stop = 1000;

figure(3);
subplot(211)
plot(1:index_stop,real(nco(1:index_stop)), 1:index_stop,real(signal(1:index_stop)));
ylabel('Real Part of PLL output');
subplot(212)
plot(1:index_stop,imag(nco(1:index_stop)), 1:index_stop,imag(signal(1:index_stop)));
ylabel('Imag Part of PLL output');
figure
```

```
plot(1:index_stop,integ(1:index_stop));
title('Integrator output');
figure
plot(1:index_stop,e(1:index_stop));
title('Error signal');
```

SIMULATED OUTPUT WAVEFORMS

SIGNATURE OF LAB IN-CHARGE

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EXPERIMENT NO-14
FREQUENCY SYNTESIZER

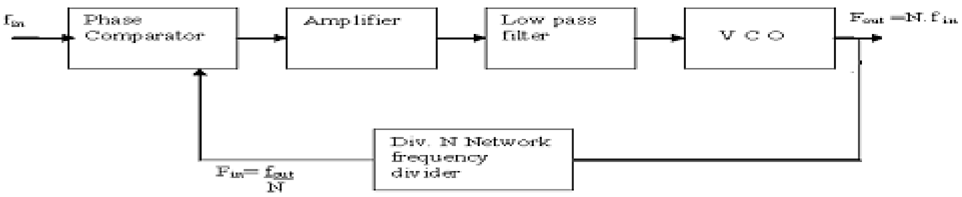
AIM: To study the operation of frequency synthesizer using PLL

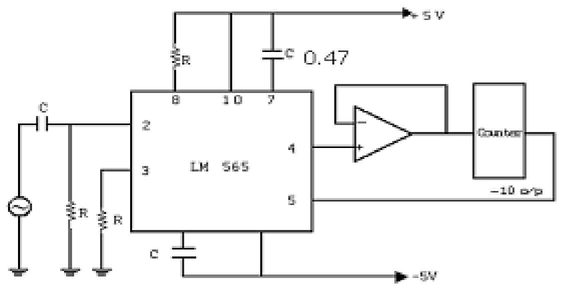
APPARATUS:

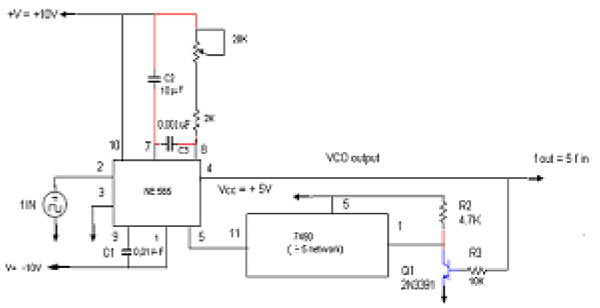
1. Frequency synthesizer trainer
2. Dual trace C.R.O (20MHZ)
3. Digital frequency counter or multimeter
4. Patch chords

BLOCK DIAGRAM:

(fig. a)







(fig b)

PROCEDURE:

1. Switch on the trainer and verify the output of the regulated power supply i.e +5v. These supplies are internally connected to the circuit so no extra connections are required.

2.Observe output of the square wave generator using oscilloscope and measure the range with the help of frequency counter, frequency range should be around 1khz to10khz.

3.Calculate the free running frequency range of circuit (VCO output between 4th pinand ground). For different values of timing resistor R1(to meaure Rt switch off the trainer and measure Rt value using digital multimeter between given test points). And the frequency values in tabular 1. Fout=0.3/(R1C1) where Rt is the timing resistor and Ct is the timing capacitor=0.01Uf.

4.Connect 4th pin of LM 565 (Fout) to the driver stage and 5th pin (Phase comparator) connected to 11th pin of 7490. Output can be taken at the 11th pin of the 7490. It should be divided by the 10,2times of the Fout.

THEORY:

The frequency divider is inserted between the VCO and the phase comparator. Since the output of the divider is locked to the input frequency Fin,VCO is running at multiple of the input frequency. The desired amount of multiplication can be obtained by selecting a proper divide by N network. Where N is an integer. For example fout=5fin a divide by N=10,2 network is needed as shown in block diagram this function performed by a 4 bit binary counter 7490 configured as a divide by 10,2 circuit. In this circuit transistor Q1 used as a driver stage to increase the driving capabilityof LM565 as shown in fig.b.

To verify the operation of the circuit, we must determine the input frequency range and then adjust the free running frequency Fout of VCO by means of R1 (between 10th & 8th pin) and C1(9th pin), so that the output frequency of the 7490 driver is midway with in the predetermined input frequency range. The output of the VCO now should be 5fin.

Free running frequency (fo):

Where there is no input signal applied, it is in free running mode.

Fo=0.3/(Rt Ct) where Rt is the timing resistor

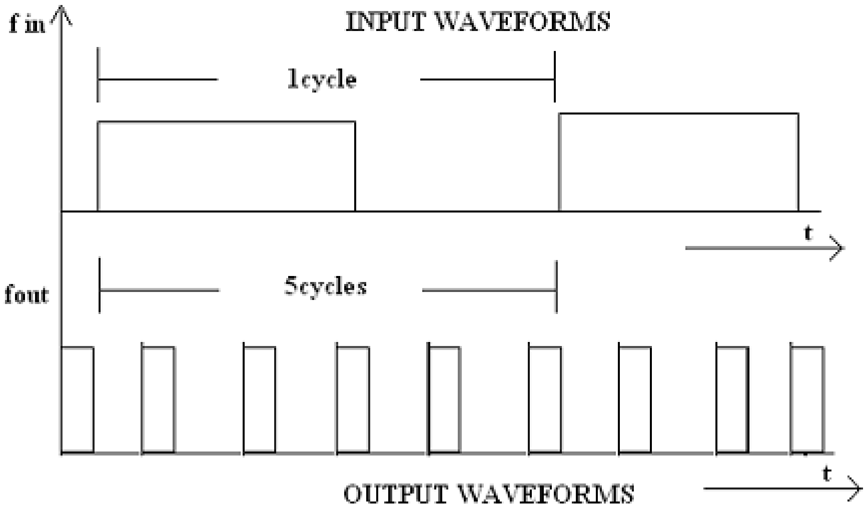
Ct is the timing capacitor.

Lock range of PLL (fl)

Fl=+8fo/vcc where fo is the free running frequency = 2Vcc

Capture range (Fc)

EXPECTED WAVEFORMS:



FinKHz	Fout = N fin KHz	Divided by 10,2

RESULT:

SIGNATURE OF THE LAB INCHARGE

QUESTIONS:

1.What are the applications of PLL?
2.What is PLL?
3.Define Lock range of a PLL?
4.What is a VCO?
5.What are the applications of frequency synthesizer?
6.What is meant by free running frequency of PLL?
7.What is the operation of a frequency synthesizer?
8.Which block is mainly used in frequency synthesizer?

MATLAB CODE:

```
%     synthesizer close all;
clear all;
fs = 10000;
t = 0:1/fs:1.5; f=50;
x1 = square(2*pi*f*t); subplot(3,1,1)
plot(t,x1); axis([0 0.2 -1.2 1.2]) xlabel('Time (sec)');ylabel('Amplitude'); title('Square wave input with freq=50HZ'); t = 0:1/fs:1.5;
x2 = square(2*pi*2*f*t); subplot(3,1,2)
plot(t,x2); axis([0 0.2 -1.2 1.2]) xlabel('Time (sec)');ylabel('Amplitude'); title('frequency multiplication by a factor of 2'); x3 = square(2*pi*f/2*t);
subplot(3,1,3)
plot(t,x3); axis([0 0.2 -1.2 1.2]) xlabel('Time (sec)');ylabel('Amplitude'); title('frequency division by a factor of 2');
```

SIMULATED OUTPUT WAVEFORMS

SIGNATURE OF LAB IN-CHARGE

ATTACH GRAPH HERE